



PROPOSTA – 3CORP_AJUSTADA_V1



DEFENSORIA PÚBLICA
DO ESTADO DO RIO DE JANEIRO

DEFENSORIA PÚBLICA DO ESTADO DO RIO DE JANEIRO (DPRJ)

PREGÃO ELETRÔNICO nº: 90017/2024 - PROCESSO Nº e-20/001.001718/2023

3CORP
Technology

A

DEFENSORIA PÚBLICA DO ESTADO DO RIO DE JANEIRO (DPRJ)

Avenida Marechal Câmara nº 314, Centro – Rio de Janeiro/RJ CEP: 20.020-080

A/c: Sr.(a). Pregoeiro(a) e Equipe de Apoio

Ref.: Pregão Eletrônico nº: 90017/2024 Processo nº E-20/001.001718/2023

Data de Abertura: 29/07/2024 – 11:00h

Objeto: FORNECIMENTO, IMPLANTAÇÃO, OPERAÇÃO E SUPORTE DE SOLUÇÃO INTEGRADA E CONVERGENTE DE ORQUESTRAÇÃO DE MÚLTIPLOS CANAIS DIGITAIS DE COMUNICAÇÃO PARA A PRESTAÇÃO DE SERVIÇOS DA CENTRAL DE RELACIONAMENTO COM O CIDADÃO DA DEFENSORIA PÚBLICA DO RIO DE JANEIRO, POR 24 MESES

CARTA PROPOSTA

Prezados Senhores,

Através deste documento, a 3CORP têm como objetivo fornecer informações referente ao FORNECIMENTO, IMPLANTAÇÃO, OPERAÇÃO E SUPORTE DE SOLUÇÃO INTEGRADA E CONVERGENTE DE ORQUESTRAÇÃO DE MÚLTIPLOS CANAIS DIGITAIS DE COMUNICAÇÃO PARA A PRESTAÇÃO DE SERVIÇOS DA CENTRAL DE RELACIONAMENTO COM O CIDADÃO DA DEFENSORIA PÚBLICA DO RIO DE JANEIRO, POR 24 MESES

Colocando-nos à inteira disposição para quaisquer esclarecimentos adicionais que se fizerem necessários.

Este documento é de responsabilidade da 3CORP não sendo permitido o uso sem autorização prévia e por escrito.

04.238.297/0001-89

**3CORP TECHNOLOGY
INFRAESTRUTURA DE TELECOM LTDA**

Alameda Oceania, Nº 56,
Polo Empresarial
Tamboré - CEP: 06.543-308
Santana de Parnaíba - SP

1. SOBRE A 3CORP

A 3CORP é uma empresa brasileira com 20 anos no mercado e é voltada para a entrega das melhores e mais avançadas soluções de Infraestrutura de TI & Telecom. A 3CORP atua como Value Added Partner da Huawei Enterprise, Premium Business Partner da Alcatel-Lucent Enterprise, Microsoft Gold Partner na competência Communications, parceira Enghouse Networks, Hikvision e Vocale Solutions.

A empresa possui uma gama diversificada de clientes nos segmentos financeiros, de governos, indústrias, hotéis, hospitais, serviços e conta com uma equipe de profissionais altamente qualificados nas áreas comercial, técnica e de desenvolvimento voltadas a criar, prover, implementar e dar suporte de acordo com a necessidade dos clientes.

Com atuação nacional, a 3CORP está com sua sede em Alphaville, Santana de Parnaíba, SP, onde funciona toda parte de Logística, Centro de Distribuição, Laboratório Técnico, Network Operations Center (NOC) e Administração Geral. A empresa conta também, com unidades em Brasília e Rio de Janeiro.

2. OBJETIVO

SERVIÇO: FORNECIMENTO, IMPLANTAÇÃO, OPERAÇÃO E SUPORTE DE SOLUÇÃO INTEGRADA E CONVERGENTE DE ORQUESTRAÇÃO DE MÚLTIPLOS CANAIS DIGITAIS DE COMUNICAÇÃO PARA A PRESTAÇÃO DE SERVIÇOS DA CENTRAL DE RELACIONAMENTO COM O CIDADÃO DA DEFENSORIA PÚBLICA DO RIO DE JANEIRO, POR 24 MESES.

3. DOS SERVIÇOS

A 3CORP Technology é um parceiro Alcatel-Lucent - “Expert Business Partner” em Business Telephony Unified communications, Collaboration/Conferencing, bem como “Value Added Partner da Huawei Technologies” e possuindo as certificações e capacitações necessárias para implementação e suporte da solução proposta.

04.238.297/0001-89

3CORP TECHNOLOGY
INFRAESTRUTURA DE TELECOM LTDA

Alameda Oceania, Nº 56,
Polo Empresarial
Tamboré - CEP: 06.543-308
Santana de Parnaíba - SP

4. PROPOSTA COMERCIAL - ANEXO IV

APRESENTAÇÃO DA PROPOSTA DE PREÇOS:

Declaramos, em atendimento ao previsto no LOTE 1 do Edital, o preço referente ao fornecimento de solução de Call Center e Plataforma Omnichannel, na modalidade SaaS (Software como Serviço), contemplando toda a infraestrutura de TI em nuvem necessária para a disponibilização da solução; Fornecedor de canais digitais de comunicação por WhatsApp Business Enterprise API, bem como soluções de bots e réguas de comunicação. Aquisição de hardware do tipo Gateway e serviço de Instalação, Manutenção e Suporte técnico dos equipamentos. Fornecedor de equipamentos do tipo Headset e adaptador de áudio para o atendimento telefônico. Banco de horas de consultoria, Serviço de treinamento técnico e de uso, Serviço Continuado de Sustentação, Garantia e Suporte e o serviço de operação assistida.

Os preços cotados são fixos, reajustáveis de acordo com os critérios do item "REAJUSTE CONTRATUAL" do Edital e expressos em Real, estão incluídas todas as incidências tributárias, despesas diretas ou indiretas, encargos fiscais, previdenciários, comerciais, sociais, trabalhistas, seguros, descontos e quaisquer outros necessários ao fiel e integral cumprimento do Contrato e seus Anexos.

LOTE 1 - Solução de Call Center

LOTE 1 - Solução de Call Center								
Item	CATSER	Descrição do produto ou Serviço	(A) Qtd.	(B) Pagamento	(C) Meses	(D) Preço Unitário	(E) Preço Mensal (AxD)	Preço Total (Cx E)
1	Solução de Call Center na modalidade SaaS (software como serviço) com hospedagem em nuvem (cloud)							
1.1	24333	Locação de licenças de uso de software de Call Center (PBX-IP Virtual) com até 350 (trezentos e cinquenta) posições	1	Mensal	24	R\$ 15.900,00	R\$ 15.900,00	R\$ 381.600,00
Subtotal								R\$ 381.600,00
2	Equipamentos							
2.1	24333	Gateway com capacidade para até 16 (dezesseis) canais E1s	1	Único	-	R\$ 81.300,00	R\$ 81.300,00	R\$ 81.300,00
2.2	24333	Instalação e Configuração do Equipamento	1	Único	-	R\$ 4.500,00	R\$ 4.500,00	R\$ 4.500,00

2.3	24333	Fornecimento e manutenção de equipamento do tipo adaptador de áudio USB	até 300	Único	-	R\$ 65,00	R\$ 19.500,00	R\$ 19.500,00
2.4	24333	Fornecimento e manutenção de equipamento do tipo Headset	até 600	Único	-	R\$ 105,00	R\$ 63.000,00	R\$ 63.000,00
Subtotal								R\$ 168.300,00
3	Outros Serviços							
3.1	24333	Banco de Horas de Consultoria	até 400	Único	-	R\$ 91,00	R\$ 36.400,00	R\$ 36.400,00
3.2	24333	Serviço Continuado de Sustentação. Garantia e Suporte	1	Mensal	24	R\$ 300,00	R\$ 300,00	R\$ 7.200,00
3.3	24333	Treinamento de usuários gestores da solução de call center para até 20 pessoas	1	Único	-	R\$ 800,00	R\$ 800,00	R\$ 800,00
3.4	24333	Treinamento técnico do Gateway para até 4 pessoas	1	Único	-	R\$ 800,00	R\$ 800,00	R\$ 800,00
3.5	24333	Serviço de Operação Assistida	1	Mensal	1	R\$ 7.800,00	R\$ 7.800,00	R\$ 7.800,00
Subtotal								R\$ 53.000,00
Valor Total da Proposta								R\$ 602.900,00 (seiscentos e dois mil e novecentos reais)

SERVIÇO PÚBLICO ESTADUAL PROPOSTA DETALHE			ANEXO II Licitação por Pregão Eletrônico nº 90017/24, A Realizar-se: 29/07/2024 - 11:00H Processo nº E-20/001.001718/2023					
A firma ao lado mencionada propõe fornecer à DPRJ, pelos preços abaixo assinalados, obedecendo rigorosamente às condições estipuladas constantes do EDITAL.			04.238.297/0001-89 3CORP TECHNOLOGY INFRAESTRUTURA DE TELECOM LTDA Alameda Oceania, Nº 56, Polo Empresarial Tamboré - CEP: 06.543-308 Santana de Parnaíba - SP					
LOTE	ITEM	CATSER	ESPECIFICAÇÃO	UNID.	QTD	MARCA/MODELO	PREÇO (R\$)	
							UNIT.	TOTAL

1 - Solução de Call Center	24333	1.1	ITEM 1 - SOLUÇÃO DE CALL CENTER NA MODALIDADE SAAS (SOFTWARE COMO SERVIÇO) COM HOSPEDAGEM EM NUVEM (CLOUD) Locação de licenças de uso de software de Call Center (PBX-IP Virtual) com até 350 (trezentos e cinquenta) posições VALOR UNITÁRIO R\$: 381.600,00 VALOR TOTAL R\$:381.600,00	Serviço	1	ATTIMO Multisserviços	R\$ 381.600,00	R\$ 381.600,00
		2.1	ITEM 2 - EQUIPAMENTOS Gateway com capacidade para até 16 (dezesesseis) canais E1s VALOR UNITÁRIO R\$: 81.300,00 VALOR TOTAL R\$: 81.300,00	Gateway entregue	1	Khomp KMG 3200	R\$ 81.300,00	R\$ 81.300,00
		2.2	Instalação e Configuração do Equipamento VALOR UNITÁRIO R\$: 4.500,00 VALOR TOTAL R\$: 4.500,00	Serviço	1	3CORP	R\$ 4.500,00	R\$ 4.500,00
		2.3	Fornecimento e manutenção de equipamento do tipo adaptador de áudio USB VALOR UNITÁRIO R\$:65,00 VALOR TOTAL R\$:19.500,00	Adaptador entregue	até 300	Adaptador TRIKS	R\$ 65,00	R\$ 19.500,00
		2.4	Fornecimento e manutenção de equipamento do tipo Headset VALOR UNITÁRIO R\$:105,00 VALOR TOTAL R\$: 63.000,00	Headset entregue	até 600	Felitron Epko Compact	R\$ 105,00	R\$ 63.000,00
		3.1	ITEM 3 - OUTROS SERVIÇOS Banco de Horas de Consultoria VALOR UNITÁRIO R\$: 91,00 VALOR TOTAL R\$:38.400,00	Hora/Pessoa	até 400	3CORP	R\$ 91,00	R\$ 36.400,00
		3.2	Serviço Continuado de Sustentação, Garantia e Suporte VALOR UNITÁRIO R\$: 7.200,00 VALOR TOTAL R\$: 7.200,00	Serviço	1	3CORP	R\$ 7.200,00	R\$ 7.200,00
		3.3	Treinamento de usuários gestores da solução de call center para até 20 pessoas VALOR UNITÁRIO R\$:800,00 VALOR TOTAL R\$: 800,00	Serviço	1	3CORP	R\$ 800,00	R\$ 800,00

3.4	Treinamento técnico do Gateway para até 4 pessoas	Serviço	1	3CORP	R\$ 800,00	R\$ 800,00
	VALOR UNITÁRIO R\$: 800,00 VALOR TOTAL R\$: 800,00					
3.5	Serviço de Operação Assistida	Serviço	1	3CORP	R\$ 7.800,00	R\$ 7.800,00
	VALOR UNITÁRIO R\$: 7.800,00 VALOR TOTAL R\$: 7.800,00 VALOR GLOBAL R\$: 602.900,00					
Observações 1ª A PROPOSTA DETALHE deverá: - ser preenchida integralmente por processo mecânico ou eletrônico, sem emendas e rasuras; - conter os preços em algarismos por extenso, por unidade, já incluídas as despesas de fretes, impostos federais ou estaduais e descontos especiais. 2ª – O proponente se obrigará, mediante devolução da PROPOSTA DETALHE, a cumprir os termos nela contidos. 3ª – A licitação mediante PROPOSTA DETALHE poderá ser anulada no todo, ou em parte, de conformidade com a legislação vigente. 4ª – Caso o Termo de Referência contenha anexo específico relativo à Proposta Detalhe, este deverá ser apresentado juntamente com o presente Anexo II.		Prazo de entrega: De acordo com o Termo de Referência Validade da Proposta - Detalhe: preços válidos por 60 (sessenta) dias úteis. Local de entrega/execução: conforme o Termo de Referência. Declaramos inteira submissão ao presente termo e legislação vigente. Em 29/ Julho/2024				
DATA: 29 / Julho/ 2024		RODRIGO ROSARIO CAVALCANTE:28364 615866 Assinado de forma digital por RODRIGO ROSARIO CAVALCANTE:28364615866 Dados: 2024.07.29 16:43:26 -03'00'				

SERVIÇO PÚBLICO ESTADUAL PROPOSTA DETALHE			ANEXO II Licitação por Pregão Eletrônico nº 90017/24, A Realizar-se: 29/07/2024 - 11:00H Processo nº E-20/001.001718/2023			
A firma ao lado mencionada propõe fornecer à DPRJ, pelos preços abaixo assinados, obedecendo rigorosamente às condições estipuladas constantes do EDITAL.			CARIMBO DA FIRMA			
ITEM	NÚMERO DE ESTOQUE	ESPECIFICAÇÃO	UNID.	QTD	PREÇO (R\$)	
					UNIT.	TOTAL
		DADOS BANCÁRIOS /LICITANTE: Banco: Banco do Brasil - 001 Ag.: 3348-0 Conta Corrente nº: 61868-3 INFORMAÇÕES PARA FATURAMENTO: DEFENSORIA PÚBLICA DO ESTADO DO RIO DE JANEIRO – DPRJ CNPJ: 31.443.526/0001-70 INSC. ESTADUAL: ISENTO	1	und.	R\$ 602.900,00	R\$ 602.900,00

END: Av. Marechal Câmara, 314 – Centro,
Rio de Janeiro/RJ CEP: 20.020-080.**Observações**

1ª A PROPOSTA DETALHE deverá:

- ser preenchida integralmente por processo mecânico ou eletrônico, sem emendas e rasuras;
- conter os preços em algarismos por extenso, por unidade, já incluídas as despesas de fretes, impostos federais ou estaduais e descontos especiais.

2ª – O proponente se obrigará, mediante devolução da PROPOSTA DETALHE, a cumprir os termos nela contidos.

3ª – A licitação mediante PROPOSTA DETALHE poderá ser anulada no todo, ou em parte, de conformidade com a legislação vigente.

4ª – Caso o Termo de Referência contenha anexo específico relativo à Proposta Detalhe, este deverá ser apresentado juntamente com o presente Anexo II.

Prazo para execução: Conforme ANEXO I.

Validade da Proposta - Detalhe: preços válidos por 60 (sessenta) dias úteis.

Local para execução: Conforme Edital e ANEXO I.

Declaramos inteira submissão ao presente termo e legislação vigente.

Condições comerciais:**Validade da proposta:** 60 (sessenta) dias, conforme edital**Prazo de entrega e instalação:** conforme edital**Local de entrega:** conforme edital**Condições de pagamento:** até 30 (trinta) dias, conforme edital**Garantia:** 60 (sessenta) meses, conforme edital**Vigência do contrato:** 24 (vinte e quatro) meses, conforme edital**Dados do representante legal da empresa para assinatura do Contrato**

Nome: RODRIGO ROSÁRIO CAVALCANTE

Data de Nascimento: 02/09/1978

Nacionalidade: brasileiro

Cargo: Diretor

Profissão: Analista de Sistema

Estado Civil: Solteiro

CPF: 283.646.158-66

RG: 25.573.598-4 SSP/SP

e-mail: governo@3corp.com.br

Telefones: 11 3056-7735/11 4118-2700

Endereço (domicílio): Rua Silva Correia, 165 – apto 132 - Vila Nova Conceição – São Paulo/SP – CEP 04537-040

Nome: GILBERTO ZÁCARO JUNIOR

Data de Nascimento: 21/12/1964

Nacionalidade: brasileiro

Cargo: Diretor

Profissão: Administrador

Estado Civil: Casado

CPF: 043.669.268-65

RG: 13.189.904-1SSP/SP

e-mail: governo@3corp.com.br

Telefones: 11 3056-7733/11 4118-2700

Endereço (domicílio): Praça Oiapoque, AP 503 - Alphaville Industrial - Barueri - SP, 06454-060



Santana de Parnaíba, 29 de Julho de 2024

RODRIGO ROSARIO
CAVALCANTE:28364
615866

Assinado de forma digital por
RODRIGO ROSARIO
CAVALCANTE:28364615866
Dados: 2024.07.29 16:43:54 -03'00'

RODRIGO ROSÁRIO CAVALCANTE - SÓCIO - DIRETOR

RG: 25.573.598-4 SSP/SP

CPF: 283.646.158-66

**3CORP TECHNOLOGY INFRAESTRUTURA
DE TELECOM LTDA**

UC & OMNICHANNEL
INFRAESTRUTURA
NETWORK & DC

04.238.297/0001-89

3CORP TECHNOLOGY
INFRAESTRUTURA DE TELECOM LTDA

Alameda Oceania, Nº 56,
Polo Empresarial

Tamboré - CEP: 06.543-308

Santana de Parnaíba - SP

Alphaville | Santana de Parnaíba - SP (Matriz)

☎ (11) 3056-7733 (11) 4450-6010

📍 Al. Oceania, 56 | Colinas da Anhanguera (Alphaville)
Santana de Parnaíba | CEP: 06543-308

Brasília-DF

📍 SCN Quadra 5 BL. A, Asa Norte,
Brasília Shopping and Towers, Ala Sul, Salas 811/813
Brasília/DF, CEP: 70715-900

www.3corp.com.br



Adaptadores

Conheça as diferenças antes de comprar



Adaptador USB

O adaptador de áudio digital USB pode ser vendido separadamente (apenas o adaptador), ou em conjunto com o headset (adaptador + headset).

Para conexão em computadores, associado a um softphone, permite o tráfego de voz através de redes de computadores, também conhecido por Voip (Voz sobre IP).

Sua arquitetura, composta por um chip controlador de áudio USB com todos os módulos essenciais incorporados, incluindo conversor DAC para Recepção e ADC para o microfone, regulador interno e transceptor USB, oferecem melhores recursos, tais como:

Aplicável em PC e Laptop

Conexão do Fone através de jack RJ9, compatível com headsets Zox e de outros fabricantes

Plug and Play, compatível com Win 98 Se, Win Me, Win 2000, Win XP, Win 7, Win 8 e Win10

Controle de volume digital no adaptador e In-line (via console do mixer multimídia)

Sigilo na transmissão (mute de gravação)

Operação e Sigilo de transmissão (mute de gravação) com indicadores luminosos (LED)

Excelente para VOIP, chat e telefonia via internet (Skype, MSN, Softphones, etc.)

Cabo USB com comprimento de 1,8m, com proteção EMI;

Uma grande vantagem em comprar este conjunto, é que qualquer headset com conexão RI-09 é compatível ao adaptador.

Pode-se deixar o adaptador fixo ao PC, e cada usuário poderá colocar seu headset individual de uso, seguindo as leis de call-center da Anatel.

Caso aconteça qualquer problema no headset, ele pode ser enviado separadamente para assistência técnica e o usuário pode colocar outro fone reserva.

Quero comprar



Adaptador Multimídia

Adaptador para uso em placa de som do computador com carona. Possui cabo com 1,80 mts. De comprimento com plug P2 injetado com diferenciação de cores conforme padrão em placa de som.

🛒 Quero comprar



Módulo de Treinamento Carona | Carrapato de Headset

O módulo de treinamento é conhecido em algumas regiões como Carrapato de Headset.

Qualidades:

O Módulo de treinamento carona é ideal para o treinamento de operadores e atendentes de telemarketing. Durante a conversação o operador ouve a ligação do supervisor para aprendizado e também o supervisor ouve a ligação do operador para orientação.

O Plug RJ9 é conectado no telefone headset e nas duas entradas RJ9 conectamos 2 headsets, onde um deles fala e ouve e no outro apenas ouve.

O Dispositivo para treinamento ou supervisão módulo carona assemelha-se a um benjamim, onde podemos conectar 2 headsets em uma base onde o 1º Headset fala e ouve normalmente e o 2º headset somente ouve a ligação. O som emitido pelos headsets deve ser menor pois o áudio será dividido entre os dois headsets.

Importante:

Módulo carona (carrapato) é compatível somente com headsets de conectores RJ-09.

Compatíveis com as seguintes marcas de headsets: Felitron (Base S8010), Zox Telecom (Bases TS19-A, TS20, TS21, TS40), TopUse (TU220) e Intelbrás (HSB20 e HSB50).

Características:

- Carona Analógico
- 01 plug macho RJ9 (4x4) e 2 entradas fêmeas RJ9 (4x4), sendo uma das entradas TX (transmissão) e RX (Recepção) e a outra somente RX (Recepção).

- Rabicho de 14 cm
- Cor: Preto
- 03 meses de garantia para defeitos de fabricação (garantia não cobre mau uso e não cobre transporte)

🛒 Quero comprar



(11) 3805-8510 | 3805-8610
☎ (11) 9-6248-2690



Av. Ministro Laudo Ferreira de Camargo,
113 - Salas 06 e 07 | Butantã - Jd. Peri-
Peri | São Paulo | Cep: 05537-001



contato@triks.com.br

Digite aqui sua mensagem

Entrar em contato

[button title="Chamar por Whatsapp" link="https://api.whatsapp.com/send?phone=5511962482690&text=Ol%C3%A1!" target="_blank" align="" icon="icon-whatsapp" icon_position="left" color="#0b4c03" font_color="#fff" size="2" full_width="" class="" download="" rel="" onclick=""]

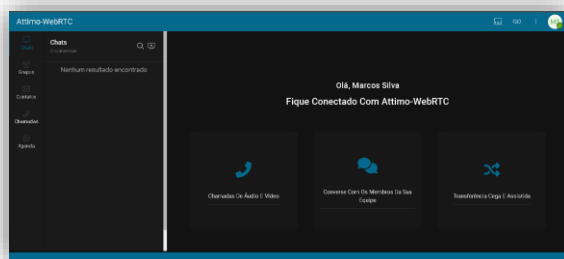
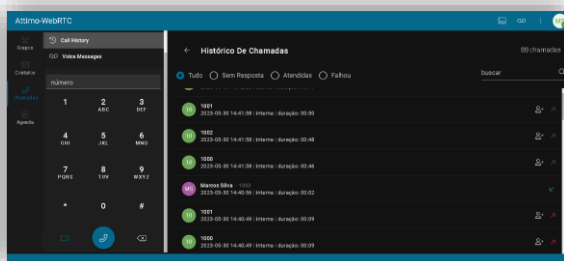


Attimo MOBILE SOFTPHONE é a ferramenta ideal para necessidades de comunicação corporativa móvel, seja dentro ou fora do ambiente empresarial seguindo a necessidade de adaptação do novo espaço de trabalho. Com a opção de Softphone para iOS, Android, Windows, Mac e WebRTC o Attimo MOBILE SOFTPHONE tem total integração com a plataforma de comunicação Attimo Cloud possibilitando que os usuários possam estabelecer comunicação interna ou externa através de múltiplos dispositivos de maneira rápida, fácil e eficiente.



Lista de Funcionalidades

- ◆ Protocolo SIP(RFC 3261)
- ◆ Multidispositivos
- ◆ WebRTC
- ◆ Windows, MAC
- ◆ Android e IOS
- ◆ Chamada Interna para Ramal
- ◆ Chamada externa para Rede Pública
- ◆ Associação a DDR
- ◆ Associação a Grupo de Busca
- ◆ Captura de chamada
- ◆ Chamada em espera
- ◆ Áudio conferência
- ◆ Rejeição de chamadas
- ◆ Consulta
- ◆ Transferência
- ◆ Função Mudo
- ◆ Lista de chamadas atendidas
- ◆ Lista de chamadas perdidas
- ◆ Lista de chamadas efetuadas
- ◆ Rediscagem
- ◆ Chamada por Nome
- ◆ Diretório
- ◆ Suporte a Headset
- ◆ Vídeo Chamada
- ◆ Interface em Português-BRASIL
- ◆ Codec OPUS
- ◆ Codec G.711 (ALAW e μ LAW)
- ◆ Codec G.722, G729 A/B
- ◆ DTMF
- ◆ Provisionamento em massa

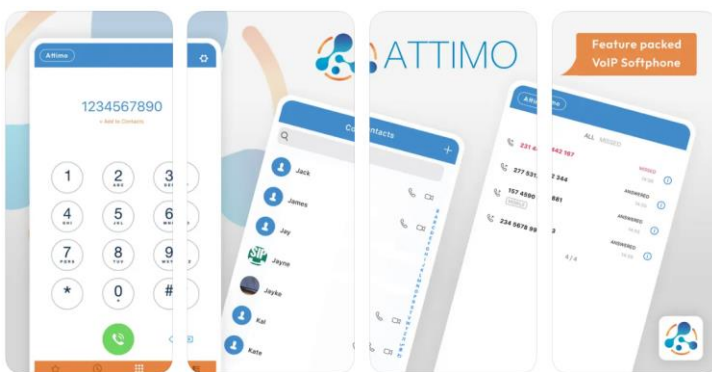
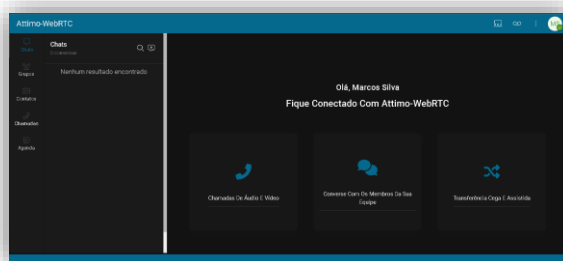
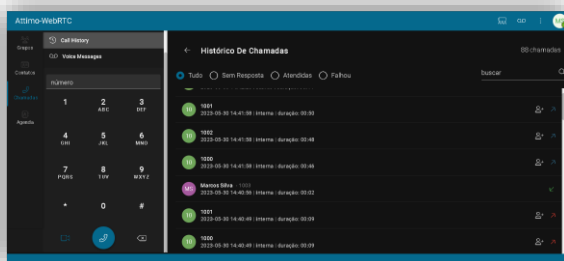


Attimo MOBILE SOFTPHONE é a ferramenta ideal para necessidades de comunicação corporativa móvel, seja dentro ou fora do ambiente empresarial seguindo a necessidade de adaptação do novo espaço de trabalho. Com a opção de Softphone para iOS, Android, Windows, Mac e WebRTC o Attimo MOBILE SOFTPHONE tem total integração com a plataforma de comunicação Attimo Cloud possibilitando que os usuários possam estabelecer comunicação interna ou externa através de múltiplos dispositivos de maneira rápida, fácil e eficiente.



Lista de Funcionalidades

- ◆ Protocolo SIP(RFC 3261)
- ◆ Multidispositivos
- ◆ WebRTC
- ◆ Windows, MAC
- ◆ Android e IOS
- ◆ Chamada Interna para Ramal
- ◆ Chamada externa para Rede Pública
- ◆ Associação a DDR
- ◆ Associação a Grupo de Busca
- ◆ Captura de chamada
- ◆ Chamada em espera
- ◆ Áudio conferência
- ◆ Rejeição de chamadas
- ◆ Consulta
- ◆ Transferência
- ◆ Função Mudo
- ◆ Lista de chamadas atendidas
- ◆ Lista de chamadas perdidas
- ◆ Lista de chamadas efetuadas
- ◆ Rediscagem
- ◆ Chamada por Nome
- ◆ Diretório
- ◆ Suporte a Headset
- ◆ Vídeo Chamada
- ◆ Interface em Português-BRASIL
- ◆ Codec OPUS
- ◆ Codec G.711 (ALAW e μ LAW)
- ◆ Codec G.722, G729 A/B
- ◆ DTMF
- ◆ Provisionamento em massa



Attimo MULTISERVICES is a Cloud Corporate Communication Platform that provides advanced IP Telephony and Mobility features for different needs and user profiles from small, medium and large enterprises. With public cloud hosting in Brazil, it follows all security, availability and confidentiality standards combined with the provision of highly reliable and simple access to communication services through any environment. Developed in line with the latest market trends such as WebRTC and Opus, ATTIMO provides flexible services that meet the requirements of new

Call Features

- Compatible with standard SIP Phones.
- WebRTC Softphone.
- Windows, Mac, iOS and Android Soft Clients
- Multi-device (5 simultaneously)
- Boss/Secretary + Whitelist.
- Blacklist.
- Busy Lamp Field (BLF) Support.
- Call Back.
- Call Flow Control.
- Call Forward (busy, no answer, unconditional).
- Call Monitoring.
- Call Pickup.
- Call Parking.
- Call Recording (Record, Search, Listening).
- Call Recording Audit and File Download.
- Call Screening.
- Call Spy.
- Call Transfer (blind, attended).
- Call Waiting.
- Callback Services.
- Caller-ID.
- 3 Party Conference Call.
- DISA (Direct Inward System Access).
- DND (Do Not Disturb).
- Direct Inward Dial Numbers (DIDs).
- Find Me/Follow Me.
- Hunt/Ring Groups.
- Music on Hold.
- Outbound/Inbound Routes.
- PIN Sets.
- Pickup Groups.
- Personal Assistance (IVR).
- Personal Recording/Notes.
- Remote Substitution.
- Remote Call Pickup.
- Route by Caller ID.
- Speed Dials.
- Three Way Calling.
- Video Calling.
- Voicemail.
- Voicemail to Email.
- Voicemail Broadcast.
- Wake-up Calls.
- Multiple Devices per User.
- Personal Call Log.
- Personal Extension Settings.
- Personal IVR.
- Voicemail to e-mail.
- Visual Voicemail.
- UC (Chat, Video and Sharing).
- User Portal: Access to Voicemail and Recordings.

Voice Mail

- Visual Indicator for Message Waiting.
- Stutter Dial tone for Message Waiting.
- Voicemail to email.
- Voicemail Groups.
- Web Voicemail Interface.
- Direct dial to operator. -

Call Center

- Agent login/log out.
- Barge.
- Call monitoring.
- Call queues.
- Call recording.
- Caller Name Lookup.
- CDR (Call Details Record).
- Conferencing (on-the-fly).
- Customer account codes.
- Hot-desking.
- Hunting groups.
- IVR / Auto-attendants.
- TTS and ASR
- Listen to agent.
- Pick-up groups.
- Queue priorities.
- Queue VIP list.
- Ring group strategies.
- Reporting.
- Time-based routing.
- Whisper to agent.

Multimedia Conference Services

- Up to 1.000 users per room
- Unlimited rooms
- Multiple Presenters
- WebRTC
- SIP Dial-in
- Audio, Video, Chat
- Conference Recording
- Screen Sharing
- Video and Audio Sharing
- Customized Background
- Noise Canceling
- Raise Hand
- White Board
- Pooling
- Video Mosaic
- Multi-party Continuous Presence
- Presenter statistics

Security

- Built-in firewall.
- Call encryption (SIP TLS, sRTP).
- AES 128, 256).
- Authorization codes.
- HMA-SHA-1
- Phone Lock.
- Intrusion detection and - blocking (Fail2ban).
- Limiting or blocking of outbound calls.
- Password strength indicator.
- PIN-protected outbound calls.
- User permission management.
- Secure password auto-generation.
- Weak password report.
- HTTPS.
- OpenVPN.

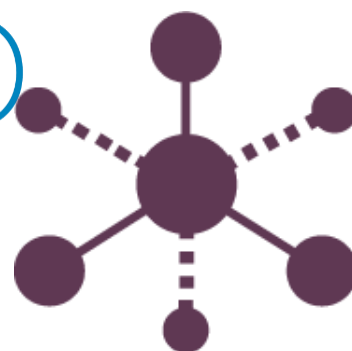
Protocols, Codecs and RFCs

- PJSIP., SIP
- IAX2.
- PRI/T1/E1/R2.
- Analog/FXS/FXO.
- ISDN.
- GSM.
- g729, g719, g722, g723.
- lpc10, slin.
- g711 A-law, g711 ulaw.
- siren14, siren7.
- g726, g726aal2.
- speex/16/32.
- adpcm, testlaw.
- gsm, ilbx, vp8, vp9.
- h261, h264, h263p, h263
- Opus
- H.264 (QCIF, CIF, VGA e HD)

Availability and Hosting

- 99,5% Availability
- Multi-DC Sao Paulo Zone
- Multi-DC Northeast Zone
- Huawei Cloud ISO 27001
- Tier 3 Certified DC
- Multi-Carrier Connection

Attimo is a trademark registered by
Attimo Telecom
Alameda Oceânia, 56 - Colinas da
Anhanguera, Santana de Parnaíba - SP,
06543-308
www.attimotelecom.com.br



Multisserviços



A **plataforma de Multisserviços ATTIMO** promove em uma única interface WEB todos os recursos necessários para operacionalização de uma central de **Contact Center** na abordagem **OmniChannel** agrupando os diversos canais de atendimento (Voz, Vídeo, Chat, Rede Social (Facebook, Twitter), Mensageria instantânea (Telegram, WhatsApp, Facebook Messenger, SMS), E-mail, Formulário Web).

A interface unificada, além de suportar múltiplas mídias, suporta Voz sobre IP em ambiente WEB, Windows, Mac além de ser possível a integração com terminais SIP e Apps para Android e iOS.

A plataforma suporta também terminais e softphones IP – Proprietário.

Através dos protocolos SIP e H.323 a plataforma suporta a integração de softphones open source.

Além dos canais de comunicação, a plataforma é composta pelos módulos de Front-End (Console do atendente, supervisor, monitor, retaguarda), Distribuidor Automático de Chamadas (DAC), Unidade de Resposta Audível (URA) com suporte às tecnologias Reconhecimento de Voz (ASR), Texto to Speech (TTS), Discador Automático, Servidor de FAX, Gravação de Voz e Tela, Bilhetador e Tarifador Financeiro, recurso de Música em Espera e publicação de mensagens, Sistema de Força de Trabalho (WFM), Módulo de Monitoria de Qualidade, Módulo de Integração Computer Telephony Integration (CTI), Atendimento Autônomo (Robô) com inteligência cognitiva e atendimento de linguagem natural, Recurso URA Visual Mobile para smartphones IOS e Android, Módulo de Gestão de Atendimento Multicanal, Relatórios Web em Tempo Real e Históricos entre outros.

A plataforma utiliza o padrão **SIP 2.0** totalmente aderente a RFC 3261, portanto em acordo as especificações da norma adotada pela Agência Brasileira de Telecomunicações ANATEL

PROTOCOLOS, SUPORTE E DESENVOLVIMENTO

- Comunicação via voz utiliza do protocolo H.323 ou SIP Padrão (RFC 3261), SIP TRUNK Padrão, RTP, RTCP e SDP.
- Comunicação com storage e aplicações: HTTP, HTTPS, FTP, SFTP, SOAP.
- Transações com sistemas: SOAP, MQSeries, ISO 8583 (Por integração). Gerenciamento: SNMP V2 ou superior.
- Acesso de gerenciamento ou atualização de correções: SSH, HTTP, HTTPS, FTP, SFTP.
- Gestão de grupos e usuários: LDAP, MS-Active Directory. Gestão de e-mails: IMAP4, SMTP, POP3.

O sincronismo de mídia e dados via CTI será principalmente via consumo WebServices, sendo possível também integrar de forma personalizada via JSON, stored procedure, JavaScript, acesso via script, plugins, arquivo texto onde disponibilizamos API e SDK documentadas facilitando assim integrações com aos mais diversos sistemas legados (CRM's/ERP's / ITSM's / Sistemas Corporativos, Provedora do serviço de e-mail e Borker SMS).

A plataforma suporta ferramenta de construção de front-end (framework de desenvolvimento) que possibilita construção de interfaces (front-end) e permite a integração com fontes de dados externos para disponibilizar informações de negócios no front-end.

A utilização de protocolo QoS para classificação de tráfego embora não necessário em função ao baixo consumo de banda principalmente quando considerado uma rede LAN.

Autenticação dos usuários ocorre através do SSO (single sign on) ou mesmo consultando base Active Directory e ou LDAP.

- Implementa e suporta os padrões IPv4 e IPv6 no mínimo os seguintes padrões:
- RFC 2460 – IPv6 Specification
- RFC 2474 - Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers
- RFC 2710 - Multicast Listener Discovery (MLD) for IPv6

- RFC 3319 - Dynamic Host Configuration Protocol (DHCPv6) Options for Session Initiation Protocol (SIP) Servers
- RFC 4291 - IP Version 6 Addressing Architecture
- RFC 4443 - Internet Control Message Protocol (ICMPv6) for the Internet Protocol Version 6
- RFC 4477 - DHCP: IPv4 and IPv6 Dual-Stack Issues
- RFC 5118 - Session Initiation Protocol (SIP) Torture Test Messages for Internet Protocol Version 6 (IPv6)
- RFC 6157 - IPv6 Transition in the Session Initiation Protocol (SIP)

Possibilita ainda a arquitetura de Serviços Multimídia sobre rede IP de forma a possibilitar a entrega de aplicações multimídia para dispositivos móveis, conforme RFC 4083 R.

Deve implementar no mínimo os seguintes padrões de comunicação:

- RFC 2543: Session Initiation Protocol v2
- RFC 2833: RTP Payload for DTMF Digits, Telephony Tones and Signals
- RFC 3261: Session Initiation Protocol
- RFC 3262: Reliability of Provisional Responses in SIP
- RFC 3263: Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264: SDP Offer/Answer Model
- RFC 3265: SIP-specific Event Notification
- RFC 3311: SIP UPDATE Method
- RFC 3323: SIP Privacy Mechanism
- RFC 3515: SIP REFER Method
- RFC 3842: SIP Message Waiting
- RFC 4028: Session Timers in SIP
- RFC 4566: Session Description Protocol
- RFC 4733: Telephone Events (DTMF)

Compatível com o padrão E.164 e permite atualização compatível ao plano de

numeração telefônico brasileiro, atendendo todas as atualizações normativas e regulatórias ANATEL.

ANATEL

No âmbito ANATEL, a plataforma de comunicação ATTIMO Voip está categorizada como Equipamento para Interconexão de Rede - Categoria III, desta forma garantida à interoperabilidade das redes de suporte aos serviços de telecomunicações, sua confiabilidade das redes de suporte aos serviços de telecomunicações e garantia da compatibilidade eletromagnética e da segurança elétrica, homologado com o número Nº 1344116970.

CARACTERÍSTICAS DE SEGURANÇA

Totalmente aderente as exigências da PCI Compliance PCI-DSS (Payment Card Industry – Data Security Standard), garantindo a proteção da privacidade e confidencialidade dos dados de cartões de pagamento. Transporte de informações sensíveis como senha, de forma criptografada utilizando algoritmo Triple-DES (3DES), com chave Double Length (32) chamada de PEK (Pin Encryption Key).

O acesso ao sistema é gerenciado através de permissões dos perfis de todos os usuários de forma integrada, mantendo os cadastros atualizados, com diversos tipos de perfis, ou seja, você gerencia tudo num só lugar, utilizando o conceito de SSO (Single Sign On).

Implementa e suporta criptografia para tráfego de sinalização e de voz entre os terminais de comunicação, entre os terminais de comunicação e o controlador de chamadas e entre os terminais e gateways para rede fixa, utilizando para isso o protocolo TLS v1.2 (RFC 5246) para sinalização segura. O protocolo SRTP (RFC 3711) para criptografia da mídia nos terminais de comunicação, utilizando o padrão AES com chaves igual ou superior a 256 bits.

Ainda permite a troca de chaves criptografia do fluxo de voz a cada chamada e distribuídas através de canal criptografado.

Implementa mecanismos de proteção para uma série de tipos de ataques como o de negação de serviço (DoS): packet replay attack, gratuitous ARPs, oversized packets, SYN floods, ping floods, Smurf ou Pong, Fraggle, Jolt1, Jolt2, Teardrop packets, Finger of death, OOB Nuke, Spank, SDP PROTOS e SIP PROTOS.

Permitir ao administrador quanto a instalação de certificado digital padrão X.509, de forma a aumentar a segurança do tráfego criptografado. Suporta o modelo PKI (Public Key Infrastructure) para gerenciamento de certificados digitais.

Tudo isso detalhado em relatórios permitindo a visualização de tentativas de login inválidas a administração do sistema.

ALTA DISPONIBILIDADE

O conjunto hardware e software foi desenvolvido para o funcionamento ininterrupto (24X7x365).

Logicamente a plataforma foi desenvolvida para atender ambiente de missão crítica com alta disponibilidade chegando a 99,999%. Essa alta disponibilidade é alcançada através de servidores em regime ativo – ativo com alta disponibilidade e redundante a falhas, de forma que a paralisação de um ou mais equipamentos isolados ou combinados não interfiram de forma impactante no ambiente.

Fisicamente o hardware é composto sempre por equipamentos dualizados, com fonte de alimentação redundante e tecnologia de disco através de RAID combinando discos sólido e mecânico.

Além disso é adotado também servidores industriais padrão de mercado sem a necessidade de utilização de hardware proprietário.

Serviços disponibilizados em regime SaaS poderão estar disponíveis em datacenter com certificação Tier 3 e até em datacenter distintos, dependendo da exigência do contratante.

IDIOMA

Toda a plataforma, módulos e relatórios estão em idioma português do Brasil, sendo possível através de arquivo de dicionário a fácil conversão para inglês e espanhol.

CARACTERÍSTICAS FUNCIONAIS

A plataforma promove o gerenciamento, integração das interações de entrada e saída (blended), independente do canal de relacionamento utilizado: voz, vídeo, e-mail, webchat, formulário na Web, rede social (Facebook, Twitter), mensageria instantânea (Telegram, WhatsApp, Facebook Messenger, SMS) e aplicativo mobile.

Esse gerenciamento é realizado em interfaces 100% web de forma unificada com tratamento e alteração de todos os tipos de mídia, não havendo necessidade de instalação de qualquer tipo de software do tipo cliente-servidor, sendo compatível com os principais browsers de mercado. A solução permite o roteamento inteligente das interações, através de uma fila única universal, independente do canal de relacionamento utilizado voz, vídeo, e-mail, webchat, formulário na Web, rede social (Facebook, Twitter), mensageria instantânea (Telegram, WhatsApp, Facebook Messenger, SMS) e aplicativo mobile.

Destaca-se na plataforma a estratégia de oferecer soluções híbridas de comunicação para seus clientes, ou a possibilidade de eles decidirem qual é a arquitetura mais relevante para seus objetivos de negócios, incluindo softphone/comunicações móveis.

Para os clientes da Microsoft, o Attimo propõe uma integração com o Skype for

Business para que pudessem aproveitar todas as possibilidades oferecidas pelo nosso sistema de comunicações de nível empresarial. Agora esta integração é potencializada com a proposta do mesmo tipo de integração com o Microsoft Teams.

A plataforma é segmentada por serviços, onde cada administrador e supervisor pode acessar somente os dados, monitoração e supervisão dos quais este usuário tem privilégio além de estar protegido por níveis de senha que garantem a integridade e segurança das informações. Os níveis de acesso e definição de senhas é definido pelo administrador Master da plataforma.

Através da interface web é possível a criação de regras de roteamento de interações com base em regras de negócio, momento e contexto da interação por qualquer canal de atendimento suportado.

Para trabalhos em equipes de atendimento, os históricos de chat e outras mídias de atendimento podem ser transferidas ou compartilhadas para outros membros da equipe.

Atendimentos multi-zona e multi-horário permitem diferentes customizações e regras de atendimento, além de personalização de diferentes saudações de voz e chat de acordo com fuso horário, horário de expediente, feriados, etc.

A plataforma suporta a criação ilimitada de scripts de tratamento de chamadas e regras de roteamento em fila. Estes scripts podem ser facilmente configurados com pequena quantidades até centenas de regras e passos.

De forma a garantir um correto funcionamento de todos os recursos ATTIMO assim como bases e sistemas externos, a plataforma é sincronizada com a Hora Legal Brasileira.

PABX IP/DAC/CTI

O PABX IP da Plataforma Multisserviços foi projetado para instituições de qualquer porte, transformando digitalmente seus negócios. É uma solução completa que inclui ferramentas de comunicação e colaboração.

Flexível, seguro e abrangente, este produto pode ser implantado em uma nuvem privada (off ou on premises) ou ainda pode ser instalado em ambientes de Collocation. A Plataforma Multisserviços é perfeita para quem busca agilidade operacional e flexibilidade, incluindo os benefícios de custos, de negócios a transformação digital contínua da empresa, essa transformação digital requer que soluções inteligentes e modernas como o Attimo sejam escolhidas.

Seguem algumas das funcionalidades e recursos da plataforma PABX IP:

- Discagem por ramal para chamadas internas e via VoIP.
- Estas chamadas podem ser aplicadas independente da região geografica.

- Suporte a serviços de FAX através de T.38 e envio e recebimento de DTMF através da RFC 2833.
- Suporte a serviço de áudio via rede pública deve utilizar o Codec G.729.
- Suporte ao Codec H.264 com resolução para video de CIF, VGA e HD 720p
- Desvios incondicionais, por ocupação, por não atendimento e indisponibilidade.
- Chamada em espera.
- Audioconferência com até 25 participantes.
- Transferência de chamadas.
- Suporte a Identificação do número chamador, Identificação do número conectado. Restrição de identificação do número de origem (Número de A), Estacionamento.
- Bloqueio de chamadas recebidas internas, externas e a cobrar por terminal. Através do tronco ou ramal.
- Bloqueio de chamadas originadas LDN, LDI, de chamadas para celular e de Chamadas para 0300, 0500 e 0900 por terminal.
- Bloqueio de chamadas, recebidas internas, internas e a cobrar por terminal.
- Grupos de atendimento, configurados de acordo com a necessidade.
- Grupos de atendimento com toques ciclico, paralelo ou sequencial.
- Quando um ramal do grupo estiver ocupado. A chamada é transferida para o próximo ramal da série.
- Retorno de chamadas.
- Log de chamadas.
- Rediscagem do último número.
- Suporte a 32 classes de serviço.
- Através das classes de serviço, será possível configurar categorias para chamadas 0300, 0800, emergenciais, essenciais, serviços, fixo local, fixo longa distancia, móvel local, movel longa distancia e internacionais.
- Chamada em espera.
- Atendimento automática URA.
- Não perturbe.
- Configurações de políticas.

- Encaminhamento de chamadas, chamada em espera, transferência acompanhada e transferência surda.
- Suporte a supressão de silêncio.
- Siga-me para numeros internos e externos.
- Suporte a armazenamento, importação e exportar de cópias das informações de configurações críticas, incluindo informações de autenticação, de forma criptografada, para a proteção dos dados.
- Possibilidade de uma única base de configuração, independentemente do número de sítios, de maneira que todas as funcionalidades e recursos devam estar presentes e disponíveis em quaisquer pontos da rede.
- Suporte ao protocolo SRTP (Secure Real-Time Protocol) para a criptografia e autenticação.
- Suporte a chaves de criptografia do fluxo de voz, possibilitando a troca delas a cada chamada entre os ramais, e distribuição por um canal também criptografado utilizando SSL/TLS para a distribuição de chaves e permissão para a implementação de certificação digital.
- Suporte a criptografia de dados fim a fim, possibilitando o tráfego seguro das informações de conversação.
- Suporte a Transport Layer Security (TLS) ou Secure Sockets Layer (SSL) para a troca de chaves simétricas e seguir o Advanced Encryption Standard (AES), de no mínimo 256 bits, padrão de mercado, para criptografia de voz durante uma chamada de Voz sobre IP.
- Suporte a utilização de chave simétrica para cada sessão e a troca de chaves deve ser realizada sempre utilizando TLS ou SSL.
- Criptografia para hardware e Software

O PABX IP DA Plataforma Multisserviços também possui integração natural com o DAC e Sistema de Gravação de Voz e Tela, Sistemas de URA, Discador automático, bem como, com as facilidades do Atendimento Omnichannel e comunica-se com as Posições de Atendimento – teleatendentes por “voz sobre IP”.

Permite através de qualquer terminal telefônico (fixo ou móvel) e validação de acesso por senha, a escuta de ligações em tempo real.

Permite a transferência automática das chamadas entre skills, ramais e caixa de mensagem, destinadas a um ramal, quando este estiver ocupado. As chamadas são roteadas como base em Automatic Number Identification - ANI - e Dialed Number Identification Service - DNIS.

Permite modificações das condições de roteamento (limitação filas de determinado

grupo e script de eventos) em tempo real e a qualquer momento antes do atendimento, através de interface gráfica. Tais limitações são realizadas de modo automático através de parâmetros específicos no aplicativo, sendo estes, dentre outros, chamada com maior tempo de espera, tamanho da fila, agentes ociosos, período da chamada, tempo médio para as respostas, chamadas prioritárias e clientes em rechamadas.

A distribuição de chamadas obedece aos seguintes critérios de roteamento cuja parametrização pode ser estabelecida de modo concomitante entre eles:

- a) especificidades e habilidades, incluindo produtividade, de cada operador.
- b) tempo de ociosidade de cada operador.
- c) quantidade de operadores ociosos.
- d) tempo de resposta.
- e) chamada em espera há mais tempo.
- f) volumetria por fila.
- g) prioridade da chamada (chamada preferencial).
- h) mês, dia, e horário da chamada.
- i) opção de atendimento na URA.
- j) tempo estimado para atendimento.
- k) prioridade para rechamadas.
- l) programação de pausas pré-determinadas pela operação.
- m) quantidade de operadores em atendimento.
- n) condições operacionais do sistema.
- o) na quantidade de agentes alocados em um grupo/especialidade.

E na quantidade de agentes disponíveis para atendimento em um skill. Permite a configuração de regras específicas (parâmetros) de roteamento para feriados e outras datas a serem definidas pela operação, sem limite de antecedência, de modo simples e prático.

Permite através de parametrização a alocação de agentes de um mesmo skill em espaços físicos distintos, possibilitando, dentre outras coisas, o ganho em escala para centrais cujas bases operacionais possuem descentralização geográfica. As características dos operadores alimentam a formação dos skills de modo dinâmico, permitindo que, em tempo de operação, sejam processadas alterações que promovam dinâmica e eficácia para alcance dos resultados operacionais.

Permite a formação de grupos (skills) disponibilizada através de acessos específicos

compatibilizados com a estrutura hierárquica de cada operação, assim um operador não tem acesso a essa funcionalidade que é executada por um superior hierárquico. As informações do operador, incluindo número de login pessoal, seguem a identificação de login independentemente da localização física do terminal de voz.

O roteamento de chamadas é administrado em alinhamento de tempo com a operação pelos gestores operacionais, sendo o acesso a essas funcionalidades parametrizáveis e, portanto, delegável pela superior hierárquico da operação.

Permite configurar o tempo de espera para atendimento estimado em bases matemáticas e o sistema disponibiliza essas informações aos clientes. Essa estimativa é dinâmica, sendo atualizada (dia, hora, minutos, etc...) por amostragem cujos parâmetros são configuráveis e assim os clientes recebem informações de estimativa de tempo atualizadas, caso seja necessário.

Quando da espera há interface harmônica de sons ou de reprodução de arquivos de áudio disponibilizados às chamadas em fila de espera. É possível a utilização de múltiplas fontes de música e diferentes anúncios para chamadas em espera, bem como regras para a repetição de anúncios.

Permite parametrizar indicações, a serem cadastradas pelos supervisores e que apontem para determinação dos tipos de chamadas, bem como o sistema possibilita que através de configuração seja obrigatória essa marcação pelos atendentes ao término de cada atendimento. Os supervisores possuem também a funcionalidade de se disponibilizarem para receber chamadas durante os períodos de maior movimento da operação.

O softphone da plataforma possui propriedades para comunicações padronizadas, sendo 100% compatível com a Plataforma e seus recursos (PABX IP, CTI). A partir deles são controlados eventos como por exemplo, login, logout, pausas, realização de chamadas e atendimento diretamente no computador da sua posição de atendimento, sendo personalizado por grupo de atendimento/especialidade, o que possibilita a integração com os diferentes níveis de acesso aos sistemas

informatizados e ainda possui funcionalidades de transferência, retenção, chaves programáveis, mudo e conferência. A plataforma suporta os codecs, para atendimento de chamadas de voz, G729 e G711.

Serviços comumente utilizados em centrais de atendimento fazem parte da gama de recursos do softphone: LogIn/ LogOff, Atendimento Interno, Atendendo Externo, Agente Livre, Pausa Produtiva, Consulta, Retenção, Chamadas de saída para a rede pública originadas pelo operador, chamadas de emergência, consulta ao supervisor, etc.

A plataforma está apta a prover capacidade de integração CTI, compondo um ambiente automatizado, o que possibilita as equipes de operação, em tempo de atendimento, consulta de informações sobre o usuário através da integração com os sistemas informatizados, assim, quando do ingresso de uma nova chamada no atendimento, o aplicativo direciona ao sistema informatizado os dados necessários para a localização do usuário.

Tais consultas ocorrem de modo síncrono possibilitando que o atendente recebe as informações o usuário e a ligação telefônica ao mesmo tempo, o que se traduz em eficiência e redução de despesas operacionais. Tais benefícios de integração se estendem à Unidade de Resposta Audível – URA, garantindo também consultas ao sistema informatizado do Call Center, por meio de webservices ou DLL, o que possibilita ações de atendimento automatizadas através das informações coletadas, favorecendo a retenção deste canal de atendimento cujos custos operacionais são consideravelmente mais baixos quando comparados o teleatendimento humano.

O sistema gerencia todos os atendimentos em todos os níveis, possuindo integração com a distribuição das chamadas e as posições de atendimento e demais níveis da operação, permitindo assim a monitoração, online, de todos os atendimentos realizados, por meio de escuta local ou remota, para fins de treinamento supervisionado conforme permissões de ramais e logins.

Além disso o sistema opera em modo multisite, possibilitando visões de apenas uma central ou visões consolidadas de várias centrais descentralizadas que compõem uma mesma operação. É disponibiliza- do um front-end para o usuário em plataforma web, com layout responsivo.

O sistema disponibiliza relatórios com informações de monitoramento do sistema e históricos ocorridos na operação da Central de Atendi- mento, de maneira local ou remota, bem como disponibiliza vários tipos de gráficos como, por exemplo, gráficos coloridos, em barras verticais, disco, etc., ou outros formatos, de maneira que demonstrem o anda- mento do atendimento em todos os níveis, bem como o status dos acordos de nível de serviço a serem alcançados e informações quanto ao tráfego de ligações telefônicas nas operações, auxiliando sobretudo na reversão de cenários negativos o mais rapidamente possível.

Além destes são disponibilizados relatórios sobre o estado do atendi- mento dos agentes e demais níveis e grupos operacionais, eventos, chamadas e navegação, em tempo real, relatórios de históricos e consolidado.

Todos os relatórios possuem recursos de agendamento para emissões futuras informando parâmetros de tempo (data e horário), bem como de customização para as mais diversas necessidades, considerando dentre outras: Login, grupos, ramais, troncos, seja para dados passados ou presentes, em tempo real, além disso possuem interfaces com recursos de alarmes visuais e sonoros, para que osusuários possam interpretá-los sem grandes dificuldades, bem como sua emissão é possível em formato como, HTML, Excel, Word, PDF, dentre outros.

VIDEOCONFERÊNCIA

Vá em frente, converse por vídeo com toda a equipe. Na verdade, convide todos que você conhece. O Attimo Instant Video Conference é uma solução de videoconferência totalmente criptografada você pode usar o dia todo, todos os dias.

O que mais você pode fazer com o Attimo Instant Video Conference?

- Compartilhe sua área de trabalho, apresentações e muito mais.
- Convide usuários para uma conferência por meio de um URL simples e personalizado.
- Escolha códigos de acesso individualizados para cada reunião.
- Compartilhamento de telas.
- Chat durante a videoconferência.
- Pedir a palavra durante a reunião.
- Gravação.
- White board.
- Estatísticas de conversação.
- Opção de votação.
- Interface compatível com Browsers WebRTC.

Uma plataforma para organizar suas reuniões. Inclui todas as funções que conhecemos das grandes ferramentas de conferência.

- Planeje sua reunião
- Proteja sua reunião com credenciais de login de usuário
- Crie um relatório de cada usuário visitando sua conferência
- Crie uma enquete de compromisso e transfira-a para uma conferência com um clique.

GRAVAÇÃO DIGITAL

Sistema de gravação digital IP para Voz, Tela e Chat simultaneamente. Está totalmente integrado à solução **ATTIMO Manager**.

Este sistema de gravação por software oferece maior confiabilidade e segurança, sigilo nas informações e transparência para auditoria, 100% aderente a Lei do SAC - Decreto 6523. Através da funcionalidade específica do módulo de Supervisão WEB

são recuperados os áudios gravados de forma fácil e intuitiva.

As buscas podem ser realizadas por definição de campos chaves a exemplo do tipo de atendimento (receptivo / ativo), código de identificação do agente / operador, número de origem do telefone que fez a chamada, etc. Os arquivos são gerados em formato não proprietários como gsm, wav e wav49 (arquivos de áudio) com taxa de compressão na proporção equivalente a 10 minutos de diálogo para cada um 1MB de espaço em disco e MP4 e AVI (arquivos de vídeo) com taxa de compressão na proporção equivalente a 10 minutos de diálogo para cada um 10MB de espaço em disco.

É possível, portanto armazenar anos de conversações e diálogos em uma mesma mídia.

Todo o armazenamento das informações poderá ser feito em equipamento especializado do tipo storage de dados.

Capacidade para a gravação simultânea de todas as chamadas recebidas, originadas, transferidas internamente e internas por grupo/fila de atendimento, agente ou ramal, de forma configurada, independe ao canal utilizado seja voz, vídeo, e-mail, webchat, formulário na Web, rede social (Facebook, Twitter), mensageria instantânea (Telegram, WhatsApp, Facebook Messenger, SMS) e aplicativo mobile.

O acesso aos relatórios de gravação se dá através da ferramenta Report (módulo de relatórios da plataforma ATTIMO)

Possui capacidade de gravação de tela do Agente de atendimento, sob demanda, no momento do atendimento realizado pelo Agente de atendimento.

Possui capacidade de armazenamento online das gravações realizadas em todos os ramais/agentes da plataforma, em mídia eletrônica, por um período de 2 (dois) anos ou por período indeterminado (offline) desde que armazenados em mídias removíveis (CD, DVD, BlueRay, DLT, AIT, LTO e DDS, etc.).

Possui capacidade de compactação das gravações nos formatos "WAV", "MP3", "GSM", "AVI", "MOV" podendo a gravação ser resgatada e reproduzida nos players de áudio existentes no mercado.

Possui capacidade de recuperação e busca em ambiente web, local ou externo.

GERENCIAMENTO DO SISTEMA

A plataforma Multisserviços possui sistema integrado de gerenciamento grafico completo, interface GUI, que oferece uma visão unificada da rede de comunicação.

O sistema de gerenciamento de rede inclui notificações instantaneas de alarme,

monitoramento de desempenho, performance, backup e restore, tudo a partir de um unico conjunto de aplicativos abrangente.

Sobre o gerenciamento do PABX virtual, a aplicação dispõe de de diferentes módulos

de administração via web, possibilitando a configuração, ampliação, supressão, entre outros recursos.

A solução disponibiliza um numero ilimitado de acessos ao portal web, com diferentes níveis de administração. Destacam-se o nível administrador, que possui total gestão da plataforma e nível visualização, que possibilita convigurações sem maiores impactos, como alteração de nomenclatura do ramal, mudança de catergoria, configuração de grupoos, entre outros..

O sistema de gerenciamento possui um módulo de inventario, com controle e registro de dispositivos, possibilitando o acompanhamento e fiscalização.

Destaca-se os seguintes recursos do sistema de gerenciamento do sistema:

- Diagnósticos internos dos módulos.
- Verificação e alteração de dados, por exemplo: - Criação de grupos, alteração de nomenclatura e categoria de ramal e/ou de tronco, alteração de número de ramal, liberação ou bloqueio de ramais, white e black liste.
- Reprogramação dos dados do sistema.
- Suporte a implementação de múltiplos fusos horários.
- Configuração da quantidade de dígitos do sistema, permitindo o bloqueio de chamadas para códigos de acesso compostos de no mínimo 8 dígitos. A inclusão de números não permitidos deverá ser realizada pelo administrador do sistema.
- Possibilidade de livre configuração de todos os recursos de telefonia, incluindo-se definição de plano de encaminhamento de chamadas, configurações de rotas, supressão de Código de Seleção de Prestadora, além de facilidades e permissões de usuários.
- Manutenção e proteção de configuração de rotas de backup, para em caso de falha na conexão com a rede. Os ramais permaneçam operacionais, e com acesso à rede pública de telefonia.
- Configuração de Rota secundaria, para em situações de ocupação de todos os canais, ou em caso de problemas que impeçam o devido encaminhamento das chamadas, elas sejam realizadas por uma outra rota, por exemplo a rede pública.
- Total controle do número de conversações simultâneas, de tal modo que nos casos em que os recursos de redes alocados para a aplicação estejam totalmente utilizados, o sistema faça encaminhamento da chamada pela rede pública ou envie sinalização de inacessibilidade de rede, não comprometendo assim a banda disponível e a qualidade de voz das ligações em curso.

- Alta-disponibilidade, o gerenciamento do sistema pode ser replicado em 02 ou mais localidades.
- Permitir manutenção remota do sistema.
- Manutenção de Gateways IP – Proprietários e Gateways SIP.
- Suporte a auto provisionamento dos telefones IPs.
- Implementação do gerenciamento via protocolo SNMP.
- Gerenciamento de múltiplos sítios por uma única interface de gerenciamento.
- Acesso via WEB ou por Software cliente servidor ou por meio de APIs públicas, tais como SNMP e SOAP.
- Visualização e conectividade dos troncos IP e o status de falhas.
- Visualização de status do dispositivo, sistema de alarmes e assistência para isolamento de problemas.
- Gerenciamento de Backup.
- Execução de Backups e Restore de configuração de todos os equipamentos da solução, excetuando-se os telefones IPs.
- Geração de relatórios da qualidade de voz nas ligações.
- Detecção de erros de programação, seja programada, manual ou periódica. A correção dos erros poderá ser realizada via aplicação de patches ou por rotinas automáticas.
- Suporte a MIB II.
- Suporte a ferramentas de manutenção apropriadas para telefonia IP, tais como relatórios de performance de rede (número de colisões, erros CRC entre outros), latência e perda de sinalização.
- Possibilidade de reinicialização dos telefones IP a partir da interface de administração.
- Suporte a gerenciamento sob protocolo SNMP ou SOAP/XML.
- Armazenamento automaticamente em disco rígido, ou em qualquer outro meio de armazenamento interno, todas as informações (“log” de ocorrências) para posterior utilização ou processamento, por período indeterminado.
- Mecanismos de proteção contra-ataques, além da proteção dos processos

rodando no servidor pela detecção de anomalias por comportamento.

- Suporte para integração com o ambiente Microsoft TEAMS permitindo e realização de ligações telefônicas, transferências de ramais, uso de caixas postais, participações em audio e video conferências, caixa postal personalizada, desvios de chamadas, intrusão, DISA e demais recursos.
- Autenticação obrigatória para registro de usuários que utilizam aparelhos ou softphones.
- Suporte a integração com o Active Directory (AD) através de autenticação via protocolo OICD/OAuth.
- Suporte a integração com serviços de diretório, suportando o protocolo LDAP para a base de usuários e método de autenticação com protocolo OICD/OAuth.
- Suporte a autenticação de usuários e segurança via LDAP ou RADIUS com AAA. Acessando serviços de diretório com autenticação via protocolo OICD/OAuth.
- Suporte a implementação de acesso por meio de senhas por usuário, bem como o controle e acompanhamento de ligações individuais.

BILHETAGEM E TARIFAÇÃO

A Plataforma Multisserviços possui sistema integrado de Bilhetagem e Tarifação que executa o controle do tráfego de chamadas registrando analiticamente os pulsos de saída individualmente por operação de Call Center e/ou telefonia administrativa.

Este módulo permite que seja feita uma análise do perfil das ligações mais utilizadas durante um período, facilitando a criação de rotas de saídas mais econômicas através da customização do servidor de voz.

Permite a geração de dados para sistemas de monitoramento de custos, gerando relatórios em português do Brasil, tanto para as legendas como para o conteúdo.

O acesso aos relatórios de bilhetador / tarifador se dá através da ferramenta ATTIMO Report (módulo de relatórios da plataforma ATTIMO).

Permite o cadastro de centro de custos, ramais e tarifas (fixo-fixo, fixo-móvel, móvel-fixo e móvel-móvel, locais, nacionais e internacionais) de inúmeras operadoras publica de telefonia.

Permite a contraposição das informações / relatório de bilhetador versus relatório de business map (relatório em layout próprio da operadora pública de telefonia) de forma a avaliar a fatura de telefonia.

Dispõe de programa de identificação das chamadas efetuadas por meio dos troncos unidirecionais e bidirecionais, com emissão de relatórios programáveis, contendo os seguintes parâmetros mínimos: número do assinante chamado em ligação urbana,

DDD e DDI (quando houver sinalização), número do ramal que originou a chamada, número de ligações por unidade da federação (quando for o caso), data de início e fim da chamada, hora de início e fim da chamada e duração da chamada.

Permite efetuar o envio dos bilhetes em tempo real de todas as chamadas.

Permite configurar a gravação dos bilhetes em disco rígido (memória não volátil).

A bilhetagem e tarifação pode ser realizada adotando os seguintes critérios:

- a) tarifação por tronco-chave e por ramal.
- b) Fornecimento de conta detalhada, com ligações fixo-móvel, fixo-fixos.

A plataforma disponibiliza detalhes da fatura e permite o download em formato editável apresentando o cálculo efetuado até o valor final.

A fatura de uso se aplica para controle e auditorias internas.

ATTIMO REPORT

A ferramenta de relatórios ATTIMO Report é uma solução Web de B.I. que permite buscar, tratar, analisar, monitorar, transformar e apresentar informações com independência e flexibilidade.

A ferramenta gera relatórios, trabalha com cubos OLAP, cria indicadores para decisões gerenciais, dashboards, integração com georreferenciamento, mineração de dados e integração de aplicações

Estão disponíveis componentes para execução de processos de ETL, criação de relatórios pré-formatados e garimpagem de dados (Data Mining).

Este módulo é acessado via Web e permite a navegação pelas diversas funcionalidades de gestão das operações de Call Center através da visualização em tempo real dos detalhes envolvendo a ocupação, demanda, volumes, resoluções, TMA, gravação e etc., cujo status podem ser medidos por gráficos atualizados a cada mudança identificada.

A partir do armazenamento das informações da plataforma, é possível extrair e gerar relatórios em tempo real e histórico do ciclo de vida completo das interações do cliente, independente do canal utilizado voz, vídeo, e-mail, webchat, formulário na Web, rede social (Facebook, Twitter), mensageria instantânea (Telegram, WhatsApp, Facebook Messenger, SMS) e aplicativo mobile.

Integrado a plataforma ATTIMO Manager, o controle de acesso com permissões aos usuários e perfis é feito através do recurso sso (single sign on).

É possível ainda o agendamento e distribuição destes relatórios de forma automática.

A ferramenta possui uma coleção de relatórios pré-definidos, atualmente cerca de 50

relatórios, divididos por assuntos: agente, bilhetador, campanha, discador, quality, omni, gravação, monitor, serviço e URA. todos estes disponibilizados na plataforma além da possibilidade de desenvolvimento e personalização utilizando a facilidade de cubos OLAP.

Permite a geração relatórios nos formatos HTML, Excel, Word, CSV, TXT e PDF.

A ferramenta de relatórios permite o acesso aos dados armazenados em banco de dados (plataforma aberta) através de views pré-formatadas sem que traga impacto a performance do ambiente.

Relatórios por Serviço / Campanha:

- Relatório de Contatos/Número do Chamador.
- Relatório de Contatos por Serviço / Campanha.
- Relatório de Estado do Serviço / Campanha.
- Relatório de Estado das Filas.
- Relatório de Produtividade por Serviço / Campanha.

Relatórios por Agentes:

- Relatório de Agentes.
- Relatório de Agentes Conectados.
- Relatório de Agentes Parados.
- Relatório de Agentes por Serviço / Campanha.
- Relatório de Produtividade por Agente.

Relatórios do Sistema de Gravação

- Relatório dos Canais de gravação.
- Relatório dos Canais de reprodução.

Relatórios por Canais Digitais

- Relatórios e-Mail.
- Relatório por Estado das Caixas de e-Mail.
- Relatórios Web Chat.

- Relatório de Sessões Web Chat.
- Relatório de Serviços / Campanhas Web Chat.
- Relatório de Produtividade por Agente.
- Relatório de Sessões de Web Chat simultâneos.

Relatórios de detalhamento de Agentes:

- Código do Grupo de atendimento/especialidade.
- Priorização do agente de atendimento.
- Código de login do agente.
- Nome do agente logado.
- Número do ramal.
- Estado do agente.
- Tempo total por estado do agente.
- Motivo da pausa.
- Horário do login.
- Quantidade de chamadas atendidas por agente
- Quantidade de chamadas ativas (chamada discada .ou de ramal saída) por agente.
- Tempo médio de conversação de todos os agentes do grupo de atendimento/especialidade.
- Tempo total de conversação de todos os agentes.
- Grupo de atendimento/especialidade.
- Tempo total de pausas por agentes.

Relatórios de detalhamento de Agentes por grupo ou especialidade:

- Código do Grupo de atendimento/especialidade.
- Nome do grupo de atendimento/especialidade.
- Quantidade de chamadas em espera na fila virtual.
- Tempo de espera da chamada da primeira chamada entrante na fila.

- Quantidade de agentes logados por grupo de atendimento/especialidade.
- Quantidade total de agentes de todos os grupos de atendimento.
- Quantidade de agentes em pausa.
- Quantidade de agentes em pronto atendimento.
- Quantidade de agentes em atendimento.
- Quantidade de chamadas recebidas por grupo de atendimento/especialidade.
- Quantidade de chamadas atendidas por grupo de atendimento/especialidade.
- Quantidade de chamadas em espera na fila..
- Quantidade de chamadas ativas por grupo de atendimento/especialidade.
- Quantidade de chamadas atendidas por tempo de espera na fila.
- Quantidade de chamadas abandonadas por tempo de espera na fila.
- Índice de nível de serviço por grupo de atendimento/especialidade.
- Tempo médio de atendimento por grupo de atendimento / especialidade.
- Tempo médio de espera por grupo de atendimento/especialidade.
- Tempo total de conversação por grupo de atendimento /especialidade.
- Índice de chamadas abandonadas por grupo de atendimento / especialidade.
- Quantidade de chamadas recebidas.
- Quantidade de chamadas atendidas.
- Quantidade de chamadas abandonadas.
- Tempo médio de espera do grupo de atendimento/especialidade.
- Tempo médio de espera do grupo de atendimento/especialidade.
- Tempo médio de espera do grupo de atendimento/especialidade.
- Tempo médio de atendimento do grupo de atendimento/especialidade.
- Índice de nível de serviço do grupo de atendimento/especialidade.
- Quantidade média de agentes logados por grupo de atendimento/especialidade.

Relatórios de detalhamento de Agentes por grupo de Supervisão:

- Código do Grupo de supervisão.
- Priorização do agente.
- Código de login do agente.
- Nome do agente logado.
- Número do ramal.
- Estado do agente.
- Tempo total por estado do agente.
- Estado do ramal do agente.
- Motivo da pausa.
- Horário do login.
- Quantidade de chamadas atendidas por agente.
- Quantidade de chamadas ativas por agente.
- Tempo médio de conversação de todos os agentes do grupo de supervisão.
- Tempo total de conversação de todos os agentes do grupo de supervisão.
- Tempo total de pausas por agente.

Relatórios históricos de Agentes de Atendimento:

- Código Agente.
- Grupo x Supervisão.
- Grupo x DAC.
- Grau de Serviço.
- Registro Primeiro Login e Último Logout.
- Quantidade Total de Chamadas Atendidas.
- Quantidade Média de Chamadas Atendidas.
- Índice Percentual de Chamadas Atendidas.
- Tempo Médio de Conversação.
- Tempo Total de Conversação.

- Quantidade Total de Chamadas Ativas.
- Quantidade de Chamadas em Hold.
- Quantidade de chamadas abandonadas no toque.
- Tempo Médio de Hold.
- Tempo Total de Hold.
- Tempo Médio de Atendimento.
- Tempo Logado.
- Tempo de Pausa Produtiva.
- Tempo Total de Pausa Produtiva.
- Tempo de Pausa Improdutiva.
- Tempo Total de Pausa Improdutiva.
- Maior Tempo de Espera.
- Maior Tempo de Conversação.
- Maior Tempo de Retenção (Hold).
- Índice de Produtividade por agente.
- Log das atividades do agente.
- Operador x chamador (ANI).

Relatórios históricos por Tráfego Telefônico:

- Quantidade de chamadas.
- Quantidade de Chamadas Recebidas.
- Quantidade de Chamadas Atendidas.
- Quantidade de Chamadas Abandonadas.
- Percentual de Chamadas Atendidas.
- Quantidade de chamadas em fila.
- Quantidade Total de Chamadas Ativas.
- Quantidade de Chamadas em Hold.
- Quantidade de chamadas abandonadas no toque.

- Tempo Total de Atendimento.
- Tempo Médio de Atendimento.
- Tempo da chamada excedido na fila de espera.
- Quantidade de Chamadas de saída.
- Percentual de Chamadas de saída.
- Tempo Médio de Chamadas de saída.
- Tempo Total de Chamadas de saída.
- Índice de Nível de Serviço.
- Índice de Abandono.
- Tempo de Espera em Fila.
- Tempo de Abandono em Fila.
- Quantidade de Chamadas Atendidas Transbordadas entre Grupos de Atendimento.
- Quantidade de Chamadas Atendidas por Estado.
- Tempo Médio de Atendimento por Estado.
- Quantidade de Chamadas Abandonadas por Estado.

Sobre a 3CORP

Somos uma empresa brasileira com mais de 20 anos no mercado voltados para a entrega das melhores e mais avançadas soluções de Infraestrutura de TI & Telecom. Atuamos como Value Added Partner da **Huawei Enterprise**, Premium Business Partner da **Alcatel-Lucent Enterprise**, **Microsoft** Gold Partner na competência Communications, **Hikvision** e **Vocale Solutions**. Somos uma das TOP 100 empresas de TI & Telecom no país e entre as 10 empresas que mais cresceram em 2019, no Ranking Exame.

Para maior excelência no atendimento a diversos segmentos, estruturamos nossa atuação em 3 unidades de negócios estratégicas: **3CORP Infraestrutura**, **3CORP Omnichannel e UC** e **3CORP Network & DC**. Temos em nosso portfólio uma gama diversificada de clientes nos segmentos financeiros, de governos, indústrias, hotéis, hospitais, serviços e conta com uma equipe de profissionais altamente qualificados nas áreas comercial, técnica e de desenvolvimento voltadas a criar, prover, implementar e dar suporte de acordo com a necessidade dos clientes.

Com atuação nacional, temos nossa sede própria em Alphaville, Santana de Parnaíba, SP, onde opera nossa Área de Logística, Centro de Distribuição, Laboratório Técnico, Network Operations Center (NOC) e Administração Geral, com mais de 250 colaboradores,

contando com unidades em Brasília e Rio de Janeiro.

Santana de Parnaíba - SP (Matriz)

+55 (11) 3056-7733 +55 (11) 4450-6010

Alameda Oceania, 56, Colinas da Anhanguera (Alphaville) - Santana de Parnaíba, CEP:
06543-308

Brasília-DF +55 61 3533-6701

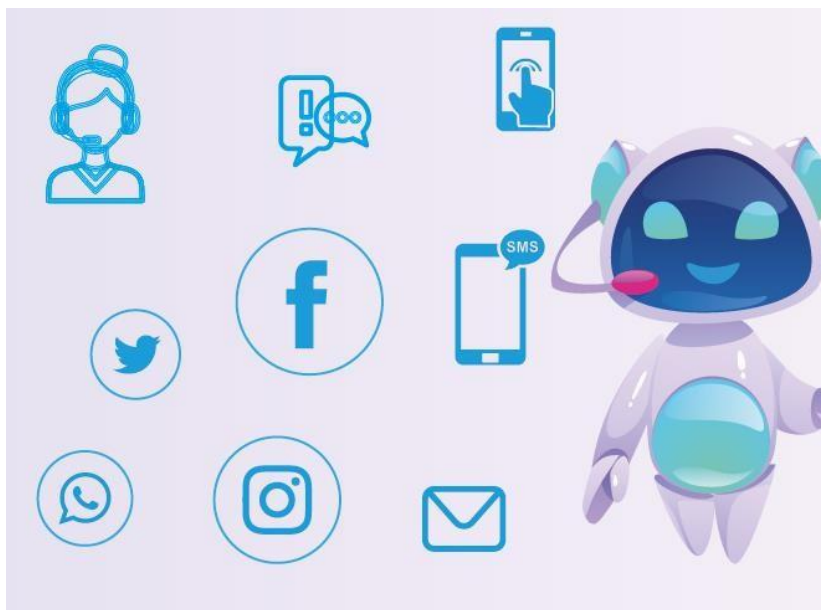
SCN QD 4 BL B Nº 100 Sala 1201, Brasília-DF - Brasília Empresarial Varig., CEP:
70.714-900

WWW.3CORP.COM.BR





Plataforma Omnichannel



A **Plataforma Omnichannel ATTIMO da 3CORP** proporciona aos clientes uma experiência consistente, independentemente do canal que eles escolherem, enquanto os agentes de atendimento têm as informações necessárias para dar suporte às interações em qualquer canal.

As soluções **ATTIMO** oferecem interações omnichannel sem interrupções que oferecem aos clientes a conveniência e a personalização que eles esperam, e aos agentes de atendimento do Contact Center, as ferramentas necessárias para o sucesso dos contatos. Todos os módulos multisserviços são integrados através da rede IP e podem ser virtualizados ou trabalharem em servidores dedicados através da rede IP 10/100/1000Base-T utilizando protocolos TCP/IP.

A estratégia Omnichannel entrega uma experiência unificada em todos os canais. Por exemplo, se um cliente interage por texto via web-Chat com o **Contact Center** e posteriormente alterna para uma chamada de voz ao telefone, qualquer agente terá o contexto completo e pode explorar onde o último agente parou.

Essencialmente, os clientes podem alternar canais e os agentes têm as informações ao seu alcance para atender facilmente os clientes, independentemente do canal.

VISÃO GERAL

A **Plataforma Omnichannel ATTIMO da 3CORP** está apta para tratar: canais de voz vídeo, e-mail, webchat, formulário na Web, FAX, rede social (Facebook, Twitter), mensageria instantânea (Telegram, WhatsApp, Facebook Messenger, SMS), integração com web sites e aplicativos móveis para serviços de “Fale Conosco, Entre em Contato” e recursos de FAX.

Outras redes sociais como exemplo do LinkedIn e Reclame Aqui são passíveis de serem integradas a plataforma, porém como que ocorre junto ao FAX, tratamos como projeto especial em função a baixa adesão dos usuários.

O tratamento de “documento físico” a partir de digitalização de ofícios e cartas também são passíveis de serem tratados porém de mesma forma dos exemplos acima é considerado um projeto especial em função a baixa adesão.

Independente ao canal de atendimento, o sistema trata todas as demandas destinadas para a central de atendimento em uma única fila universal com quantidade ilimitada de circuitos de anúncios digitais e regras e prioridades independente do canal de relacionamento utilizado. Nesta fila o sistema distribui a demanda em acordo ao perfil de atendimento do agente, skill, prioridade, habilidade ou outro destaque deste profissional.

FUNCIONALIDADES

MANAGER OMNICHANNEL

Permite o gerenciamento, integração, orquestração e tratamento de chamadas /demandas receptivas e ativas em modo blended através de interface Web amigável e intuitiva, destinada a central de atendimento por intermédio de múltiplos canais de comunicação. Estes canais são classificados no sistema como chamadas / demandas síncronas (tempo real) e assíncrona (não sensível a espera).

A quantidade de acessos ao Manager Omnichannel está limitada a quantidade de licenças de agentes atribuídas na plataforma, não sendo necessária licença específica para esta aplicação de gerência.

Possibilita a criação de política de distribuição, roteamento e priorização para interações omnichannel para todos os canais suportados pela plataforma, em fila única, com base em critérios configuráveis de: Regras de Negócio, Tempo, Data, Hora, Modo Noturno, Perfil de Agente, Perfil de Cliente, Volume de interações, Status de Fila, Tipo de Serviço, Opção de URA, dentre outros.

Permite o roteamento de e-mail e de solicitações de formulários web com o registro histórico completo da interação, bem como suporta múltiplas caixas de entrada de e-mail que podem ser agrupadas por tipo de serviço/campanha.

Permite definir e personalizar os tipos de status disponíveis (login, logout, pausa NR17, banheiro, lanche, reunião, etc.), criar e gerenciar perfis de agentes, supervisores, administrador, monitor da qualidade, back office, control desk entre outros, bem como a criação atributos, skill, grupo, tipo de serviço, filas de atendimento, criação e associação de perfil de discagem (ramal, local, móvel, DDD, DDI).

Permite a visualização de informações de agentes e grupos de agentes: livre, ocupado, atendendo chamada externa; atendendo chamada interna, pausa produtiva, pausa, retenção, consulta; realizando chamada e interação multimídia.

Permite a visualização de múltiplas interações em tempo real e relatórios estatísticos com múltiplas informações de interação incluindo número do chamador e informações coletadas na URA, assim como quantidade de chamadas abandonadas e o número do chamador que abandonou.

Permite a criação e definição de múltiplas campanhas ativas unitárias ou simultâneas de voz com ou sem URA, integradas a campanhas de e-mail e sms, estes últimos utilizando recurso/servidores próprio ou brokers de envio em massa, permitindo a sua realização no mesmo ambiente de atendimento receptivo (modo blended).

Esse modulo oferece um editor gráfico de URA que permite ao usuário (não programador) a desenvolver estruturas de navegação a partir do simples arrastar de objetos pela tela e edição destes objetos utilizados de conceito drag and drop.

A integração nativa com a URA inteligente permite a transferência de chamadas entre o fluxo da URA para o DAC e dos agentes DAC para a URA sem interrupções.

A ferramenta permite criar, editar, salvar, alterar, salvar novas mensagens, mudanças de horários, filas e grupos de atendimento sem limite de quantidades de filas e grupos.

Realiza a vinculação de mensagens de avisos visuais e ou auditivos (alarmes), adequados ao canal de interação omnichannel, permitindo ativação de mensagens padrão ou personalizada do sistema em acordo ao tipo de status do agente e ou mensagem;

Permite a integração e ou importação de bases de conhecimento existentes independente a base de dados (Microsoft SQL Server, Oracle, Postgresql, MySQL entre outros), ou a obtenção das informações via fonte de dados externo através de comunicação webservice, de forma que tal fonte poderá alimentar a inteligência do A.I. Shophia (ChatBot).

Permite que integrações com bases de dados e ou sistemas externos direcionem as interações, baseadas em regras de negócio/serviços, histórico de serviços e/ou preferências pessoais. As integrações poderão ser por meio de arquitetura orientada a serviços ou utilizar outros padrões técnicos de mercado.

Permite o desenvolvimento de workflows de atendimento através da interface Web para todos os canais disponíveis na plataforma, de forma fácil e intuitiva possibilitando aos agentes executar processos de atendimento com atividades estruturadas e informações integradas aos sistemas de negócio (CRM's, ERP's, etc.) associados ao negócio e registradas no histórico da jornada do cliente.

A interface de gerenciamento web permite criar e gerenciar a jornada do cliente baseado nos objetivos de negócio da operação, permitindo a visualização de status do agente, incluindo ausente, ocupado, disponível, pausa NR17, demais pausas ou outros status de forma personalizada.

Administradores, supervisores, control desk, monitor da qualidade, agentes de atendimento e demais papéis do contact center são definidos a partir da criação de perfis distintos e posteriormente associados ao usuário de forma a herdar os privilégios sob a plataforma.

Perfis de acesso de gestão (supervisor e agente) podem interagir através de recursos de comunicação interna seja via voz ou chat para apoio aos agentes de atendimento através de suas respectivas telas de atendimento (front-end).

A tela de atendimento do agente permite, de acordo com o skill de atendimento, a associação da fila e definição de canal, a comunicação, seja de ativa e ou receptiva, em uma mesma interface de atendimento, independente do canal utilizado.

A tela de supervisão (front-end), possui a capacidade de apoiar o atendimento do cliente participando também da fila de atendimento ou consultar o agente, monitorar, emitir sussurro, interceptar a chamada corrente, capturar a tela do atendimento corrente de forma online, interferir no status trocando status (livre, ocupa, pausa) do agente e deslogar caso seja necessário.

Permiti o gerenciamento da funcionalidade de apoio ao agente pelo supervisor definido por tipo de serviço / campanha ou grupo de login / skill.

A tela de atendimento permite que o agente apoie o cliente através do recurso de co-browse. Esse recurso permite, coma autorização do usuário, a interação do agente na tela do cliente facilitando principalmente os atendimentos com características de service-desk.

A plataforma suporta as funcionalidades de web-callback com o mecanismo de gravação das interações com o cliente e co-browse.

Todas as interações, independente ao canal de comunicação, são registradas da forma de log gravação do atendimento (voz, vídeo e chat) permitindo a recuperação do histórico do atendimento, independente ao canal de comunicação.

A recuperação do histórico de um atendimento realizado, é possível ainda que o agente possa mudar de canal de atendimento sem perder o contexto da conversa. Exemplo: Cliente entra em contato junto a central de atendimento através do Webchat. Posteriormente retorna via Whatsa- pp e finaliza o atendimento através do Telegram. A mudança entre os diversos canais de comunicação ocorre sem que haja perda de contexto da conversa.

Possibilita que uma chamada via voz, seja armazenada pela sua gravação e posteriormente transformá-la em texto através do recurso de TTS (text to speech).

A ferramenta para o tratamento da voz pode ser configurado para realizado tanto por softphone ou através da tela de atendimento por intermédio do protocolo WebRTC (Real-Time Communication) incluindo chamada de vídeo, ficando a critério da operação.

A tela de atendimento permite disparar mensagens do tipo SMS, e-mail desde que o agente possua perfil adequado para esse tipo de tarefa.

A Plataforma possibilita a implementação de roteamento de chamadas baseado no nível de serviço e tempo estimado de espera.

Suporta troncos publicos TDM, E1-CAS, ISDN, IP, H.323, SIP.

Permite integração com sistema de discador automático com geração de relatórios

AGENT OMNICHANNEL

Permite consolidar em uma única interface 100% web o atendimento entre os diversos canais de atendimento. A interface é única e configurável para cada perfil de usuário, grupo de atendimento ou mesmo tipo de atendimento/serviço e trata os canais de voz, vídeo, e-mail, webchat, formulário na Web, FAX, rede social (Facebook, Twitter), mensageria instantânea (Telegram, WhatsApp, Facebook Messenger, SMS) e aplicativo mobile.

Permite a configuração de multisessões da interface do agente de atendimento para interações receptivas e ativas de forma flexível, independente do canal utilizado, e a configuração do registro do histórico de cliente detalhado por canal.

Permite que o agente trabalhe em modo blended (receptivo e ativo), com suporte para atender diversos canais simultaneamente e acesso ao perfil do cliente quando identificado, inclusive seu histórico de interações detalhado, sejam elas receptivas e/ou ativas.

O Front-end suporta, de maneira configurável, a disposição das informações do cliente, no formato de screen pop-up, dos scripts de atendimento, dos front-end de sistemas integrados que são utilizados no apoio ao atendimento (CRM's, ERP's, ITSM's, dentre outros) de acordo com os serviços associados, o perfil do cliente e do perfil do atendente habilitado para o atendimento.

O Front-end suporta, de maneira configurável, a disposição das informações das operações de status (login, logout, pausas, intervalo, filas, etc.), da comunicação com o chamador, do registro do atendimento via lista pré-programada e texto livre, das respostas automáticas a serem inseridas pelo atendente através de FAQ, da informação do humor do cliente através do recurso de speech analytics.

Permite configurar a visualização das informações dos usuários capturadas ou identificadas automaticamente pelos diversos canais de atendimento e apresentando no próprio front-end tais informações.

Funciona em ambiente 100% web não sendo necessário a instalação de qualquer software do tipo cliente-servidor, sendo esse voltado para desktops e plataformas de mobile (Smartphones, tablets), bastando para isso um browser padrão de mercado. O funcionamento do front-end é garantido em qualquer browser padrão de mercado.

O front-end para plataforma mobile é simplificado apresentado somente o controle da operação, as informações do cliente e espaço para preenchimento das informações. Ainda em plataforma mobile o front-end poderá fazer uso do recurso de coordenadas de GPS, de forma a identificar a posição do agente.

Pela configuração do perfil do agente / supervisor, os ícones disponibilizados assim como o comportamento do front-end e seu formato serão ajustados em acordo as opções e definições de acesso e perfil de usuário (voz, vídeo, e-mail, webchat, formulário na Web, rede social (Facebook, Twitter) e mensageria instantânea (Telegram, WhatsApp, Facebook Messenger, SMS) e aplicativo mobile).

A Plataforma permite ao agente trabalhar em modo de atendimento ativo ou receptivo, sendo que, na ausência de chamadas de entrada, o sistema automaticamente deverá alocar o

agente para o sistema de chamadas de saída (outbound).

Permite a definição de fluxos com aprovação do Supervisor antes do encaminhamento do e-mail do agente ao cliente, permitindo selecionar todos ou grupos de e-mails e agentes e, ainda, definir percentual de 0 a 100% dos e-mails que deverão ser direcionados à aprovação do Supervisor.

O acesso à interface será em língua portuguesa e utilizar protocolo seguro, tal como o HTTPS, de forma a proporcionar comunicação segura, com autenticação individual por agente.

TRATAMENTO DE E-MAIL

Permite o tratamento dos e-mails entrantes ou iniciados pelos agentes nas campanhas ativas, incluindo confirmações de recebimento e respostas automáticas, bibliotecas com templates configuráveis, roteamento e enfileiramento inteligente baseados nas habilidades dos agentes e na fila de espera.

Suporta os protocolos de e-mail POP3 e IMAP.

Permite configurar a criação de filas de atendimento com distribuição

associadas às regras de roteamento com base em: informações constantes nos campos assunto e no corpo do e-mail, disponibilidade e habilidade do agente, e grupo de agentes / serviços.

Permite o recebimento e envio de e-mails em uma ou mais caixas, conforme configurado na solução, transformando as demandas em e-mail e vice-versa, conforme parâmetros e fluxos previamente definidos.

Permite a configuração de formulários diferentes para produtos e clientes.

Permite utilizar e-mail na formatação HTML de texto e imagem, permitindo configurar o envio de e-mails com arquivos anexos.

Permite a geração de mensagens automáticas para o mailing indicado ou mesmo em resposta a uma chamada realizada através de outro meio.

Possui a função de programação para enviar mensagem de recebimento para cada fila de atendimento de E-mail.

Permite configurar o envio de arquivos anexos: .pdf, .csv, .doc, .xls, e.html, txt e outros portadores de conteúdo descritivos.

Permite configurar análise dos campos: origem, destino, assunto, corpo do e-mail.

Permite configurar controle do tempo máximo que um e-mail possa ficar retido com um dos agentes.

Permite ao administrador criar “notificações automáticas” para cada fila de atendimento. A “notificação automática” é um recurso que informa ao cliente que o e-mail enviado a central de atendimento já foi recebido pelo sistema.

Permite ao administrador do sistema criar regras para descartar

e-mails (SPAM). A identificação deste e-mail deverá ser baseada nas informações contidas no título e corpo do e-mail, baseado em análises e categorização.

Permitir ao administrador do sistema a criação de cabeçalhos e assinatura de e-mails para cada fila

de atendimento do Contact Center.

A plataforma permite o atendimento de múltiplas interações e intercalação de múltiplos canais de atendimento simultaneamente, sendo configurável pelos administradores e Supervisores, com suporte ilimitado a interações simultâneas, onde o status do agente automaticamente pode ser definido como ocupado ao atingir o número de atendimentos simultâneos pré-configurado.

TRATAMENTO DE SMS

A solução permite a interação com os clientes através de troca de SMS (short message service), bem como, o gerenciamento de campanhas de envios individuais ou massivos de SMS.

Permite ao cliente receber SMS e respondê-los para a solução para tratamento na fila de atendimentos omnichannel.

Permite o gerenciamento e roteamento de SMS de entrada com base em palavras chave, identificações do cliente, data e hora, número de origem e localização.

Permitir a configuração para que o cliente SMS possa ser identificado com base no número telefônico de A (originador da chamada), devidamente registrado nos sistemas legados integrados à solução, e armazenamento das mensagens SMS por pelo menos 2 anos.

Permite encaminhar mensagem de SMS como alerta, informação, URL's, dentre outras em campanhas ou nas situações indicadas no motor de decisões ou em opções realizadas pelo Cliente, independente do meio de acesso do mesmo.

Permite encaminhar SMS ao cliente com notificações do tratamento de suas demandas/chamadas (independente do meio de acesso inicial), quando este possuir telefone celular cadastrado. Exemplo de tratamentos: registro, prorrogação de prazo, respostas, pendências, etc.

A Plataforma permite a integração com a Solução de envio de SMS, mantendo o atendimento no mesmo front-end de chat.

TRATAMENTO DE CHAT E INSTANT MESSENGER

Suporta o tratamento de chat disponibilizando aplicação "click-to-chat" a ser instalada em servidores web do contrate, visando disponibilização de interface para o cliente acessá-lo via browser comuns de mercado (IE, Chrome, Firefox, Safari etc), seja via PC, notebook, tablets ou smartphones.

Suporta o tratamento de Instant Messenger – IM (Telegram, WhatsApp, Facebook Messenger) a partir de APIs de integração dos fabricantes. Permite a configuração de autenticação e criptografia, com integração com os sistemas de segurança de terceiros, bem como suportar APIs para integração com aplicações da CONTRATANTE.

Permite identificar o status do usuário, inclusive ausência, e configurar mensagens de conforto, indicando o status do agente através de símbolos ou mensagens como "digitando" e "online".

Deve permitir que o agente visualize que o cliente está escrevendo e vice-versa.

Possui facilidade de parametrizar as interfaces web internas e externas com layout responsivo.

7 3CORP | Dataheet ATTIMO Omnichannel V2

Permite configurar que atendentes e clientes compartilhem telas e formulários com uso de cobrowse, e cooperem para preencher os formulários, vendo as mudanças feitas por cada um.

Permite que atendentes utilizem a base de conhecimento (FAQ) no tratamento da demanda e que o conteúdo da base possa ser enviado ao cliente via URL, vídeo ou texto diretamente na conversa. A FAQ permite a criação de frases frequentemente utilizadas pelo agente com acesso por menu exibido durante a sessão de chat e/ou IM.

Permite implementar método de criptografia que codifique os dados transmitidos entre o transmissor e o receptor e vice-versa, de forma a inviabilizar a monitoração por terceiros.

Permite o acesso a todo o histórico de interações com o cliente via Web Chat/IM e demais canais de atendimentos disponíveis ao perfil do agente. As mensagens podem ser armazenadas em base de dados própria da solução e estarão disponíveis para consultas por um período não inferior a 2 anos.

Permite configuração de aviso ao agente de atendimento, por sinal sonoro ou por interface de pop-up da tela de atendimento do agente quando o cliente solicita atendimento on-line, ou outra forma de destaque da solicitação que venha a disponibilizar.

Permite ao administrador criar “notificações automáticas” para cada fila de atendimento, a exemplo de sudações “bom dia”, etc. A “notificação automática” é um recurso que informa ao cliente mensagens, baseadas em análises e categorização e sem a interação de um atendente.

Permite a realização de, pelo menos, 4 (quatro) sessões de chat e/ou IM por Operador ativo.

Permite utilizar sessões com segurança SSL.

Permite configurar mensagens de informação para o cliente.

Permite configurar URLs para que o cliente possa acessar uma página enquanto aguarda na fila de espera.

Permite que o atendente inicie uma chamada de outro meio, a partir de uma chamada corrente.

Permite configurar a transferência da tela do atendimento para atendimento de 2º nível, transferindo o histórico de conversa para o atendente de 2º nível e notificando o cliente.

Permite configurar entrada de informações consideradas confidenciais que serão digitadas pelo cliente e que serão ocultadas para o atendente (exemplo: número do cartão de crédito).

Permite a criação de filas de atendimento com distribuição de sessões de web chat e IM ao agente ou grupo com base em regras de roteamento por disponibilidade e habilidade do agente.

Permite aos supervisores o recurso de “whisper”, permitindo que o supervisor visualize a conversa entre o agente e o cliente e envie mensagens visíveis apenas ao agente e Supervisor.

Permite que o supervisor monitore o agente de forma discreta, sem que o agente seja notificado sobre essa monitoração.

Permite que o supervisor, durante um monitoramento, possa intervir na conversa entre o operador e o chamador, intercalando a chamada.

Permite ao agente a seleção de status pronto / disponível (online), ocupado, desconectado / indisponível (offline) e em pausa, sendo este último customizável e permitindo a definição de motivos

de pausa.

Permite ao agente finalizar o atendimento e escolher uma tabulação pré-definida ao atendimento. Exemplo: Serviço prestado com sucesso, reclamação, etc.

Permite o atendimento de múltiplas interações de web chat e IM ou outros canais de atendimento simultaneamente sendo configurável pelos administradores e supervisores, sendo que deve ser suportado o máximo de 10 atendimentos simultâneos de web chat ou IM.

Permite aos administradores e supervisores definir a intercalação de interações de chat e/ou IM com outros canais de atendimento (voz e e-mail) habilitados ao mesmo usuário.

Permite configurar regras de roteamento e distribuição para que, quando o agente estiver ocupado, novas interações devam ser encaminhadas ao agente livre que tenha o perfil para atendimento deste canal ou ficam em fila de atendimento.

Permite configurar a transferência do atendimento para outro agente, mantendo o histórico da conversa.

TRATAMENTO DE REDES SOCIAIS

A Solução suporta o tratamento para as demandas oriundas das redes sociais Facebook e Twitter.

A solução prover os “adaptadores”, de maneira nativa, para os principais serviços de Mídia Social (Facebook, Twitter). A função dos adaptadores será de “ouvir”, monitorando as atividades das redes sociais relativas ao cliente.

Permite configurar recursos para, por meio de regras específicas, efetuar o monitoramento dos posts por meio de palavras-chave, frases, pedaços de palavras e múltiplas palavras, transformando-as em demandas internas a serem tratadas por meio de ferramenta de tratamento de demandas.

Permite configurar recursos para varredura nas redes por meio de palavras chave, frases, pedaços de palavras e múltiplas palavras. Permite efetuar a entrega das demandas a serem atendidas para grupos específicos de atendimento parametrizáveis.

Permite que o cliente seja respondido pela própria solução, sem necessidade de outros tipos de acesso à rede social.

Permite configurar recursos para configurar filtros baseados em regras de negócio.

Permite configurar o compartilhamento da mensagem na própria rede, por exemplo: retweet e compartilhamento do Facebook.

Permite configurar a contagem do número de fãs, seguidores, compartilhamentos, cliques realizados e menções aos termos monitorados nas redes sociais, separados por tema ou gerais.

Permite configurar a contagem do alcance das publicações realizadas. Permite configurar a contagem de uma nuvem de palavras dos termos mais publicados relacionados ao contratante, separados por tema ou gerais.

TRATAMENTO DE VÍDEO

A solução permite implementar acesso do cliente via vídeo-chamada ponto a ponto para tratamento na operação omnichannel.

Permite o estabelecimento da vídeo-chamada iniciadas para o cliente, em APP móveis, caso ele opte por esse contato via esse meio de acesso.

Permite configuração para que o cliente possa ocultar sua imagem e colocar-se em mudo, devendo o frontend do atendente informar tal ação ao mesmo caso parametrizado.

Permite configurar o agendamento de vídeo-chamada conforme estabelecido pelo cliente.

As vídeo-chamadas poderão ser realizadas durante um atendimento em outro canal, através de solicitação do cliente e execução por parte do atendente.

Sobre a 3CORP

Somos uma empresa brasileira com mais de 20 anos no mercado voltados para a entrega das melhores e mais avançadas soluções de Infraestrutura de TI & Telecom. Atuamos como Value Added Partner da **Huawei Enterprise**, Premium Business Partner da **Alcatel-Lucent Enterprise**, **Microsoft** Gold Partner na competência Communications, **Hikvision** e Vocale Solutions. Somos uma das TOP 100 empresas de TI & Telecom no país e entre as 10 empresas que mais cresceram em 2019, no Ranking Exame.

Para maior excelência no atendimento a diversos segmentos, estruturamos nossa atuação em 3 unidades de negócios estratégicas: **3CORP Infraestrutura**, **3CORP Omnichannel** e **UC e 3CORP Network & DC**. Temos em nosso portfólio uma gama diversificada de clientes nos segmentos financeiros, de governos, indústrias, hotéis, hospitais, serviços e conta com uma equipe de profissionais altamente qualificados nas áreas comercial, técnica e de desenvolvimento voltadas a criar, prover, implementar e dar suporte de acordo com a necessidade dos clientes.

Com atuação nacional, temos nossa sede própria em Alphaville, Santana de Parnaíba, SP, onde opera nossa Área de Logística, Centro de Distribuição, Laboratório Técnico, Network Operations Center (NOC) e Administração Geral, com mais de 250 colaboradores, contando com unidades em Brasília e Rio de Janeiro.

Santana de Parnaíba - SP (Matriz)

+55 (11) 3056-7733 +55 (11) 4450-6010

Alameda Oceania, 56, Colinas da Anhanguera (Alphaville) - Santana de Parnaíba, CEP: 06543-308

Brasília-DF +55 61 3533-6701

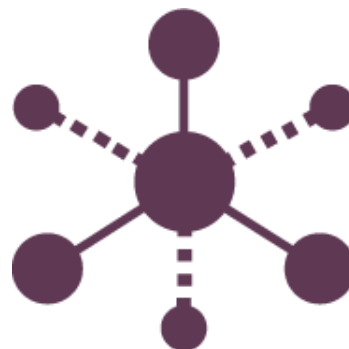
SCN QD 4 BL B Nº 100 Sala 1201, Brasília-DF - Brasília Empresarial Varig., CEP: 70.714-900

WWW.3CORP.COM.BR





URA



FUNCIONALIDADES

As Soluções de Contact Center da ATTIMO possuem plataforma de URA que permitem que clientes em atendimentos receptivos (in-bound) executem transações e obtenham informações sem ter que falar com um operador ou recorrer à atendimentos especializados de pessoal de back-office.

A plataforma também disponibiliza um módulo de URA Ativa que potencializa campanhas de contatos ativos (outbound) via sistema automático de voz. Assim que a chamada é realizada por discador automático, o sistema redireciona-a para uma URA previamente definida. Na URA Ativa podem ser disponibilizadas roteiros vocalizados pretendidos, bem como promover acolhimento de respostas dos clientes durante o atendimento automático.

Este serviço é tipicamente usado na implementação de questionários de pesquisa e de processo de CPC (contato com a pessoa certa).

A plataforma suporta virtualização ou é implementada em servidor físico conectado a rede ethernet 10/100/1000Base-T, utilizando protocolo TCP/IP;

CONFIGURÁVEL E CUSTOMIZÁVEL

A plataforma URA da ATTIMO é altamente configurável, personalizável e disponibiliza uma interface completa e intuitiva que garante uma fácil configuração de um sistema interativo e automático de voz destinado a responder

ou interagir com o interlocutor, permitindo, por exemplo, direcionar as chamadas para os operadores mais aptos num determinado assunto.

Através de interface gráfica intuitiva, é possível alterar a árvore de atendimento de forma simples e rápida, criando e editando novos fluxos e agregando novas mensagens e gravações de áudio sem interrupção nos serviços de atendimento. A alteração de fluxos pode ser feita sob demanda do cliente ou o próprio cliente pode absorver conhecimento para este tipo de alteração.

URA INTELIGENTE

Nossas soluções de URA utilizam a vanguarda dos recursos tecnológicos de Reconhecimento de Voz (ASR – Automatic Speech Recognition) e Conversão de Texto (TTS - Text-to-Speech), promovem a redução de custos operacionais e melhoram a satisfação do cliente, otimizando o tempo de contato através de autoatendimento ou por interações com assistentes virtuais (Bot's), através de plataforma cognitiva com inteligência artificial e emprego de linguagem natural, eliminando o desconforto da espera de atendimento.

INTEGRAÇÃO SEM LIMITE

O módulo de URA pode usar as informações de negócio de bancos de dados externos e aplicações diversas tais como CRM's, ERP's, ITSM's dentre outros, para personalizar os atendimentos do cliente e aumentar significativamente a eficiência de cada interação do Contact Center.

A integração com diferentes plataformas de base de dados (MS SQL, SQL, Postgres, Oracle, WebServices) traz flexibilidade ao ambiente de produção.

Através de canais SIP ilimitados em topologia multi-core, cada módulo possui suporte para 120 canais SIP para interconexão com diversas marcas e modelos de PABX, Gateways e Operadoras SIP. A topologia pode ser "In-Front" ou "Behind" ao sistema DAC.

INFORMAÇÕES TÉCNICAS

- VISÃO GERAL

Os recursos possibilitam registro das transações, considerando as aplicações, por meio do Call Detail Records – CDR, bem como configurações de objetos, edição de mensagem, gravação de mensagem, vocalização de números, reprodução de voz, coleta de caracteres, tom de chamada, sinal de ocupado, beep, menu de sobre discagem, condicional, condicional de tempo, estado do aplicativo, desvio incondicional, desvio múltiplo, temporizador, integração, via web service, com Sistemas de Gestão do Relacionamento com clientes - CRM.

FUNCIONALIDADES

- Possibilidade de acesso às informações contidas em um banco de dados, através de seleção da opção pelo teclado, no caso de aparelho telefônico tom (DTMF), disponibilizando na mesma ligação as informações desejadas, via voz em sincronismo com os dados;
- Possibilidade de atendimento com opções distintas para diferentes tipos de usuários de acordo com regras pré-definidas armazenadas em base de dados, através de referenciais como, CPF, RG, Protocolos de atendimentos, etc.;

- Capacidade de identificação dos dígitos informados, com retorno ao demandante da recepção de cada caractere digitado, por mensagem de voz ou por meio de bips ou eco quando se tratar de senhas, bem como reprodução artificial de voz através da funcionalidade TTS (Text-to-Speech) e reconhecimento de Voz (ASR – Automatic Speech Recognition) baseado no idioma português do Brasil, com possibilidade de inserção de outros idiomas como, por exemplo, o inglês. Além disso a solução de URA permite a vocalização de datas em diferentes formas como dd/mm/aaaa, dd/mm, mm/AA, dd/mm/aa, de valores monetários (Reais– R\$) encadeando adequadamente as unidades, dezenas, centenas, milhares e milhões, efetuando comparações de numerais tanto inteiros, como decimais e moedas;
- Possibilidade de configuração para vocalizações em que se faz necessário que o usuário ouça toda a mensagem, possibilitando em seguida a digitação por parte do mesmo;
- Possibilidade para que vários chamadores ouçam o mesmo anúncio de modo simultâneo;
- Disponibilidade, através de regras de negócio definidas previamente, e de modo automático, de dados e informações requisitadas no menu de navegação;
- Permissão que o usuário interrompa a sequência de informações (opções disponíveis) e escolha a opção desejada em qualquer ponto da navegação através da facilidade cut-thru;
- Possibilita sequências de atendimento diferenciados para períodos específicos pré-definidos como, por exemplo, dias, semanas, horários, etc.;
- Disponibilidade da configuração timeout, para definição do tempo máximo para aguardar a digitação de toda a informação e também do tempo máximo de espera entre a digitação de cada dígito;
- Disponibilidade da função “screen pop-up” de modo a evitar que o cliente tenha que repetir algo já informado no caso de transferência da ligação para, por exemplo, uma célula de segundo nível de atendimento ou para a supervisão, conforme o caso;
- Fluxos de serviços totalmente independentes;
- Possibilidade de alterações nas mensagens (gravação, criação e edição) e fluxos através de aplicativo de gerenciamento da URA por acesso remoto, restrito por senha.
- Possibilidade de criação, alteração e atualização de árvores de voz de acordo com cada aplicação e necessidade utilizando solução que possibilita criações e alterações a partir de módulos funcionais, por objetos, do tipo drag-and-drop, diminuindo a necessidade do uso de linguagens de programação, principalmente em atividades mais simples e sem a interrupção da operação da URA;
- Programação via building blocks das etapas de atendimento de cada fluxo;
- Possibilidade de simulação, controle sinalização de falhas;
- Possibilidade de configurações por tipo de atendimento, controles, assinante, arquivo de voz, bilhetagem, mensagens, usuário, etc.;
- A solução contempla diversas fontes de áudio para chamadores em fila de espera como, por exemplo, .WAV, MP3, AAC, etc.,
- As gravações podem ser realizadas pelos processos de digitalização em estúdio, com alta qualidade de som, bem como a partir de mensagens gravadas digitalmente em microcomputador;
- Relatórios de solução de URA acessíveis, mediante cadastro prévio e senhas de acesso, a vários usuários de modo simultâneo (mínimo de 100 usuários), sendo que para todos os relatórios estão disponibilizadas, de modo

parametrizável, opções de visualização a partir da duração das ligações (tempo mínimo, médio ou máximo), dados agrupados por ano, por mês, por dia e por faixas horárias, sendo essas a partir de intervalos de 30 minutos e opções de URA;

- Possibilidade de emissão ou agendamento de emissão de relatórios a partir de parâmetros pré-definidos (periodicidades, modelos, dados agrupados por ano, por mês, por dia e por faixa de horário, distribuição geográfica de origem das ligações, em um período selecionado, com opção de abertura por opção final na URA. etc.) para geração futura;
- Possibilidade de emissão em formatos diversos ou em tabelas e gráficos (csv, doc, xls);
- Relatório de atendimento personalizado conforme todas as opções disponíveis aos clientes na URA, tempos de utilização para cada opção, de forma agrupada ou separada, conforme parâmetro escolhido, identificando quantidades, percentuais, etc;
- Disponibilização de todas as ligações recebidas no período selecionado (data e hora inicial e data e hora final), número do telefone, cidade e estado, tipo de ligação (fixa ou móvel), data, horário e duração de cada ligação.
- Disponibilização do tipo de encerramento da chamada (usuário, teleatendente, URA). Sendo que na URA é identificado o ponto de desconexão de modo a avaliar possíveis problemas no trâmite do fluxo por parte dos usuários.
- Disponibilização da “rastreadabilidade” da ligação dentro da URA, com ordenação cronológica, resultado e tempo de atendimento em cada etapa, bem como total de ligações que entraram em cada faixa horária, sendo essas parametrizáveis a partir de intervalos de 30 minutos e identificação das perdas dentro da URA e sua representatividade tanto em relação ao total de ligações no mesmo período (também parametrizáveis), com abertura de cada um dos motivos. Também são apresentados os resultados, especificando se a opção selecionada atendeu as expectativas do demandante, bem como identificando possíveis erros.
- Disponibilização de informações sobre os horários de picos da operação de teleatendimento, com informações de volumetria, tempo médio de atendimento e nível de serviço nos períodos de maior concentração de chamadas.

RELATÓRIO PADRÕES DE URA:

- Número máximo de canais de Canais Telefônicos Ocupados.
- Número de Chamadas Telefônicas Recebidas.
- Número de Chamadas Telefônicas Finalizadas na URA.
- Representatividade de Chamadas Telefônicas Finalizadas na URA em relação ao Total de Chamadas Recebidas.
- Tempo de Navegação na URA.
- Tempo Médio de Navegação na URA
- Número de Chamadas direcionadas para o Atendimento Humano
- Representatividade de Chamadas Direcionadas para Atendimento Humano em relação ao Total de Chamadas Recebidas.
- Tempo Médio de Atendimento Humano – TMA

- Tempo Médio de Atendimento Total (Humano e URA)
- Tempo Médio de Navegação na URA.
- Ao término do atendimento humano, a Plataforma possibilita o redirecionamento das chamadas para à URA, de forma a realizar a possibilitar ao cliente resposta à pesquisa de satisfação pós-atendimento telefônico.
- A funcionalidade de pesquisa permite a criação de lógicas, em que a resposta do cliente determina as próximas perguntas.
- A plataforma permite rodar de forma ilimitada, pesquisas de satisfação simultaneamente.
- A plataforma possibilita a associação da pesquisa de satisfação à gravação e ao agente relacionado à chamada que antecedeu ela, permitindo à Contratante uma análise minuciosa das notas e da qualidade do atendimento em si.
- A plataforma suporta arquivos de voz no formato .WAV com codificação linear PCM 16-bit e 8KHz Mono com “bit rate” de 64kbts/seg ou superior.
- A plataforma suporta integrações via API REST.
- A plataforma possibilita a configuração da funcionalidade “force play” para vocalizações.

Sobre a 3CORP

Somos uma empresa brasileira com mais de 20 anos no mercado voltados para a entrega das melhores e mais avançadas soluções de Infraestrutura de TI & Telecom. Atuamos como Value Added Partner da **Huawei Enterprise**, Premium Business Partner da **Alcatel-Lucent Enterprise**, **Microsoft** Gold Partner na competência Communications, **Hikvision** e Vocale Solutions. Somos uma das TOP 100 empresas de TI & Telecom no país e entre as 10 empresas que mais cresceram em 2019, no Ranking Exame.

Para maior excelência no atendimento a diversos segmentos, estruturamos nossa atuação em 3 unidades de negócios estratégicas: **3CORP Infraestrutura**, **3CORP Omnichannel e UC** e **3CORP Network & DC**. Temos em nosso portfólio uma gama diversificada de clientes nos segmentos financeiros, de governos, indústrias, hotéis, hospitais, serviços e conta com uma equipe de profissionais altamente qualificados nas áreas comercial, técnica e de desenvolvimento voltadas a criar, prover, implementar e dar suporte de acordo com a necessidade dos clientes.

Com atuação nacional, temos nossa sede própria em Alphaville, Santana de Parnaíba, SP, onde opera nossa Área de Logística, Centro de Distribuição, Laboratório Técnico, Network Operations Center (NOC) e Administração Geral, com mais de 250 colaboradores, contando com unidades em Brasília e Rio de Janeiro.

Santana de Parnaíba - SP (Matriz)

+55 (11) 3056-7733 +55 (11) 4450-6010

Alameda Oceania, 56, Colinas da Anhanguera (Alphaville) - Santana de Parnaíba, CEP: 06543-308

Brasília-DF +55 61 3533-6701

SCN QD 4 BL B Nº 100 Sala 1201, Brasília-DF - Brasília Empresarial Varig., CEP: 70.714-900

WWW.3CORP.COM.BR





Billing

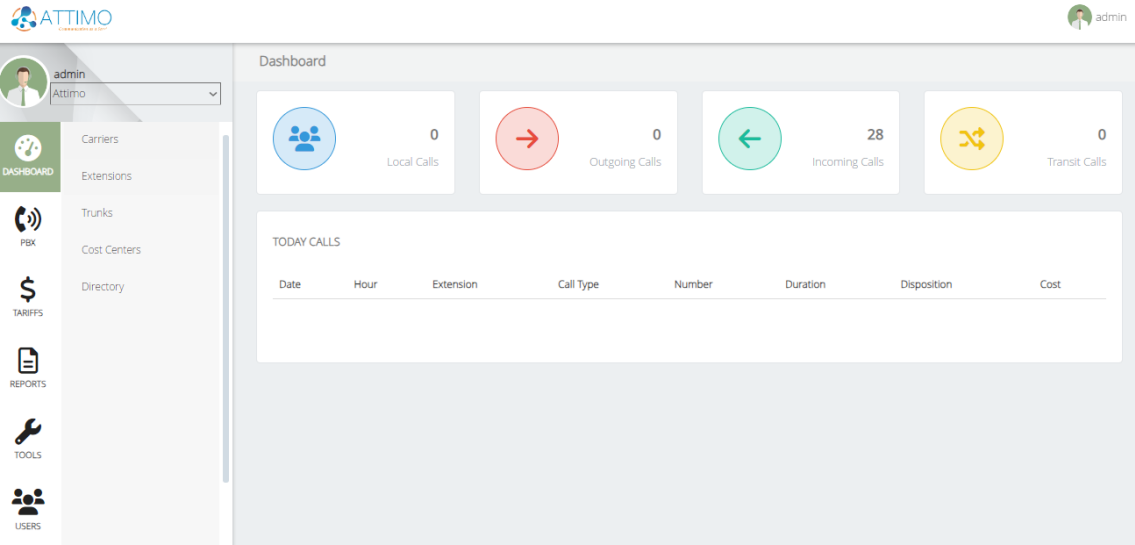
INDEX

Dashboard	3
PBX	4
Tariffs	7
Reports	12
Tools	15
Users	16
Settings	17

Dashboard

Dashboard

The Dashboard shows the last calls of the system and is updated every minute. The newest call is shown at the top.



PBX

Carrier

The first step to be able to rate a call is to define the different carriers to which we are going to interconnect, for this we must go to PBX/Carriers.

Carriers

GENERAL

Carrier

Shared ?

☐

Save

The information to be configured are the following:

Carrier, just write the name of the carrier, which will be used later.

Share, if checked, the operator and the rates assigned to it will be shared with the other tenants.

Trunk

It is necessary to synchronize and configure the trunks as these are associated with a Carrier and this way you can charge the calls more accurately. There are also reports by trunks which help us verify if what the Carrier is charging is correct. To configure the trunks, it is necessary to go to PBX/Trunks.

Trunks

GENERAL

Trunk Device

Description

Technology

PJSIP

Pilot Number

Carrier

Default

Synchronize

Save

The data to be configured are the following:

Trunk Device, the name of the trunk as it appears in the call, for example in the case of Dahdi channels it is a number.

Description, short description to identify the trunk

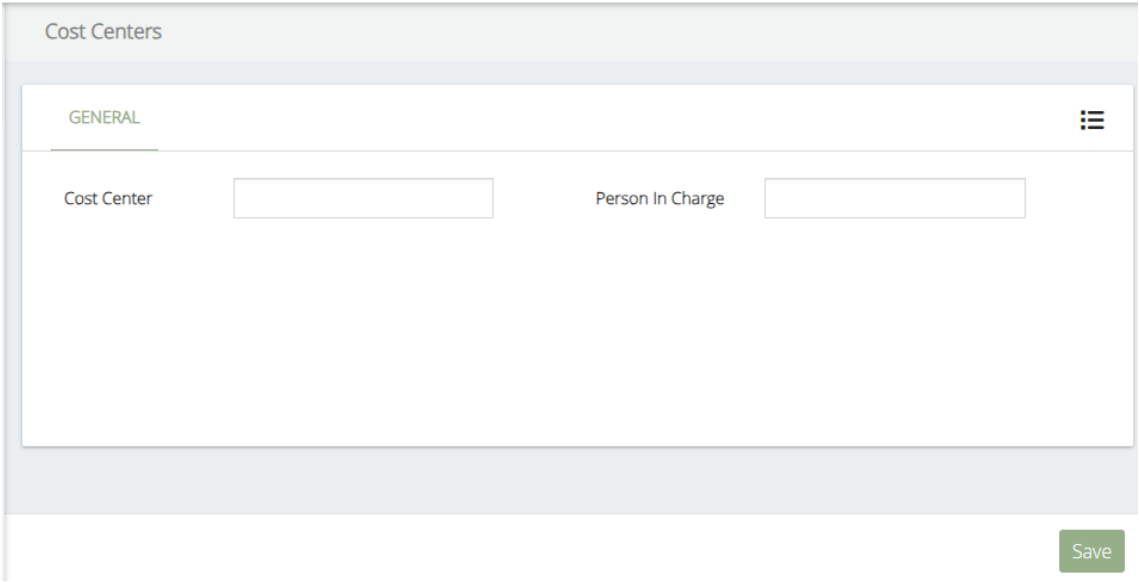
Technology, type of trunk technology

Pilot Number, number of pilot associated to the trunk, can be a DID or a telephone number that identifies the trunk.

Carrier, the carrier that provides the service. Very important as this depends on how you will charge the calls that are taken by this trunk.

Cost Center

It is possible to obtain reports organized by cost centers, for this it is necessary to assign the extensions the Cost Center to which it belongs.



The screenshot shows a web interface for configuring Cost Centers. At the top, there is a header bar with the text "Cost Centers". Below this, there is a tab labeled "GENERAL" with a hamburger menu icon to its right. The main form area contains two input fields: "Cost Center" and "Person In Charge". Below these fields is a large, empty rectangular area. At the bottom right of the form, there is a green "Save" button.

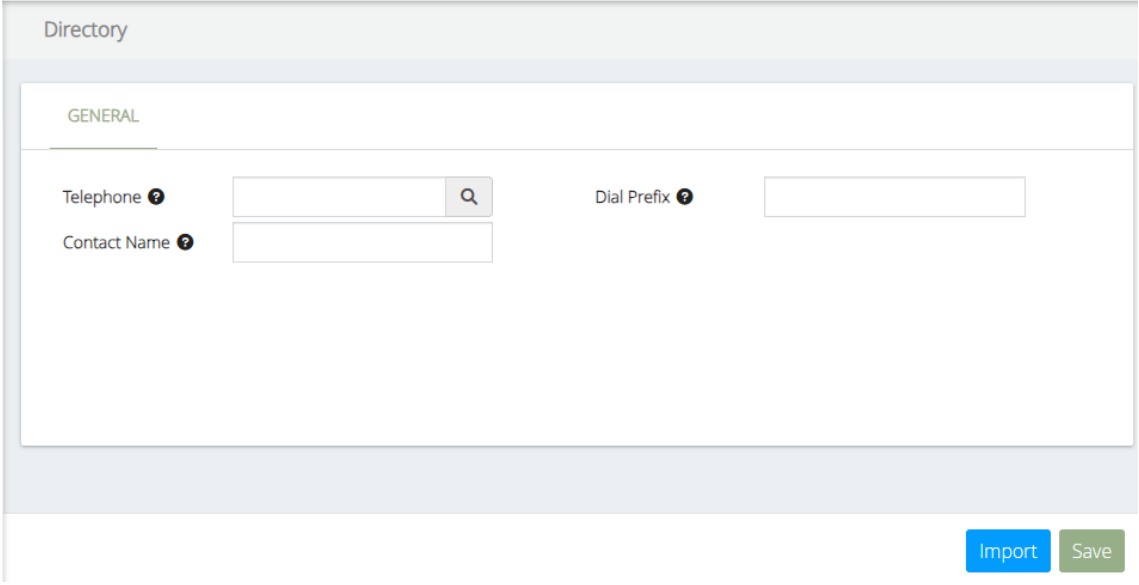
The data to be configured are the following:

Cost Center, short name of the cost center.

Person in Charge, Person in charge of the Cost Center.

Directory

In order to better identify the destinations of the calls it is possible to add an external telephone directory. In the reports will appear the number alongside with the contact name configured here.

The screenshot shows a web interface for configuring a directory. At the top, there's a header bar with the word "Directory" in a light blue font. Below this is a tab labeled "GENERAL" in green. The main area contains three input fields: "Telephone" with a search icon, "Contact Name", and "Dial Prefix". Each field has a small question mark icon to its right. At the bottom right of the form, there are two buttons: "Import" in blue and "Save" in green.

The data to be configured are the following:

Telephone, phone number to be included in the directory. Should be unique.

Contact Name, brief description of the contact or company.

Dial Prefix, when a call is made from the PBX in the CDR it is usually stored together with the prefix, for example, if we are going to call the number 88823700, and to call it is necessary to dial 9, in the CDR 988823700 will be stored. The prefix 9 must be added here to correctly identify the outgoing call. For incoming calls receiving CID this prefix is not used.

It is also possible to import a csv list that contains the following information and in the following order.

mode, add or delete.

full_name, brief description of the contact or company.

phone_number, phone number to be included in the directory. Should be unique.

dial_prefix, when a call is made from the PBX in the CDR it is usually stored together with the prefix, for example, if we are going to call the number 88823700, and to call it is necessary to dial 9, in the CDR 988823700 will be stored. The prefix 9 must be added here to correctly identify the outgoing call. For incoming calls receiving CID this prefix is not used.

Tariffs

Before connecting the program to the PBX you must first define the tariffs of the different carriers, otherwise we run the risk that the cost of calls is not correct. If for some reason we connect the program to the PBX before setting the rates, do not worry, you can reprocess the calls or at the moment of obtaining reports choose the option to “Apply Current Rates”.

Carrier Rates

It is very easy to create the tariffs of the different carriers, for this process there are two methods, which we will show below:

The screenshot shows the 'Carrier Rates' configuration page. It features a 'GENERAL' tab with various input fields for defining carrier rates. Below this is a 'Dial Codes' section with a table for listing codes, costs, and intervals. The 'Import' and 'Save' buttons are located at the bottom right of the page.

1. Import rates from csv file.

To import tariffs, you must press the Import button, then there will be two fields:

Carrier, name of the Carrier associated with the tariffs to be imported, this name must have been previously created in PBX/Carrier

Rates, load CSV file with the rates to import. This file must have the following format:

Group Name, name that the rates are grouped, example if we are going to load tariffs to USA, which places USA, in other words could be the name of the country.

Code, country or region code, example USA is 1, Mexico 52. Can also be more specific USA, Miami 1305.

Call Cost, Cost of the call.

Cost Interval, every few seconds the cost applies.

Code Description, specific description of the code, example in case of code 1305, in Group name is USA but in Code Description is Miami.

Dial Prefix, if it is necessary to add a prefix to the code, in many countries the number “00” is used for international calls.

To see an example of the CSV format you can download it by clicking the button “Download CSV Format”.

2. Create rate manually.

It is possible to create or modify an existing tariff, to create a new tariff manually you must configure the following:

Description, brief description of the group of codes to be created.

Grace Time, define the seconds will be free from the call duration for each call.

Connect Cost, cost to apply when the call began with connection.

Call Cost, represents the global cost to apply to all the dial codes without a cost defined.

Cost Interval, it defines how often seconds will be applied the global cost.

Carrier, defines to which carrier belongs this group of dial codes and rates.

Dial Prefix, any number that is dialed before a telephone number to initiate a telephone call for the purpose of selecting an appropriate telecommunications circuit by which the call is to be routed.

Prepend, any number to prepend at the beginning of the dialed number, at the time of showing a report.

Call Type, allows you to define which kind of calls these dial codes represent.

Dial Code Category, allows you to categorize your dial codes to make reports by it.

In the Dial Codes table, the following must be configured:

Code, the full or partial dialed number as was saved on the CDR.

Call Cost, represents the global cost to apply to all the dial codes without cost defined.

Cost Interval, It defines how often seconds will be applied the global cost.

Description, brief description of the code to be created.

Dial Prefix, any number that is dialed before a telephone number to initiate a telephone call for the purpose of selecting an appropriate telecommunications circuit by which the call is to be routed. If it empty, the global is taken.

Prepend, any number to prepend at the beginning of the dialed number, at the time of showing a report. If it empty, the global is taken.

Customer Rates

It is very easy to create personalized rates, this option is very useful for Hotels or institutions that charge calls, for this process there are two methods, which we will show below:

Customer Rates

GENERAL

Description

Grace Time 0

Connect Cost 0

Default Cost 0

Default Cost Interval 60

Default Dial Prefix

Call Type Outgoing

Dial Codes Category Uncategorized

Dial Codes

Code	Call Cost	Cost Interval	Description	Dial Prefix

Add New

Import Save

1. Import rates from csv file.

To import tariffs, you must press the Import button, then there will be two fields,

Rates, load CSV file with the rates to import. This file must have the following format:

Group Name, name in which the rates are grouped, example if we are going to load tariffs to USA, which places USA, in other words could be the name of the country.

Code, country or region code, example USA is 1, Mexico 52. Can also be more specific USA, Miami 1305.

Call Cost, Cost of the call.

Cost Interval, every few seconds the cost is applied.

Code Description, specific description of the code, example in case of code 1305, in Group name is USA but in Code Description is Miami.

Dial Prefix, if it is necessary to add a prefix to the code, in many countries the number "00" is used for international calls.

To see an example of the CSV format you can download it by clicking the button "Download CSV Format".

2. Create rate manually.

It is possible to create or modify an existing tariff. To create a new tariff manually you must configure the following:

Description, brief description of the group of codes to be created.

Grace Time, define how many seconds will be free from the call duration for each call.

Connect Cost, cost to apply when the call with connection.

Call Cost, represents the global cost to apply to all the dial codes without cost defined.

Cost Interval, it defines how often seconds will be applied the global cost.

Dial Prefix, any number that is dialed before a telephone number to initiate a telephone call for the purpose of selecting an appropriate telecommunications circuit by which the call is to be routed.

Prepend, any number to prepend at the beginning of the dialed number, at the time of showing a report.

Call Type, allows you to define which kind of calls these dial codes represent.

Dial Code Category, allows you to categorize your dial codes to make reports by it.

In the Dial Codes table, the following must be configured:

Code, the full or partial dialed number as was saved on the CDR.

Call Cost, represents the global cost to apply to all the dial codes without cost defined.

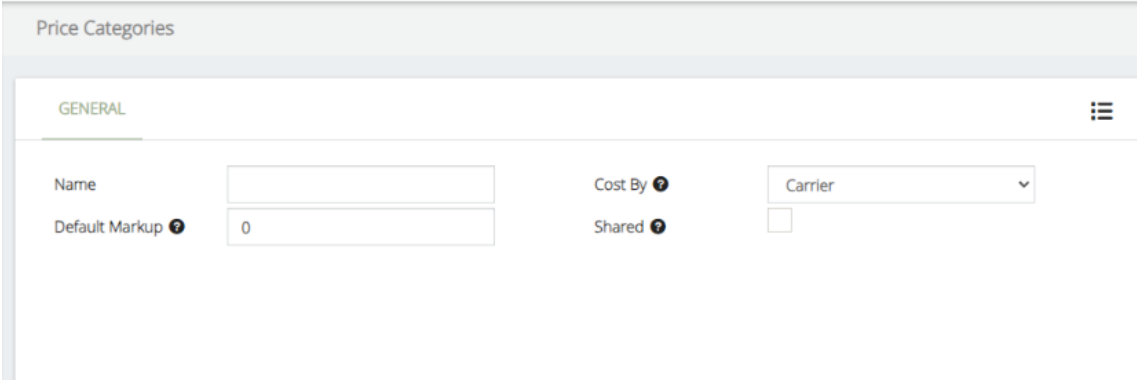
Cost Interval, It defines how often seconds will be applied the global cost.

Description, brief description of the code to be created.

Dial Prefix, any number that is dialed before a telephone number to initiate a telephone call for the purpose of selecting an appropriate telecommunications circuit by which the call is to be routed. If it empty, the global is taken.

Price Categories

Each extension is associated to a Price Category. This has the aim of being able to charge in differentiated form by extension. Very useful for Hotels.



The screenshot shows the 'Price Categories' configuration interface. At the top, there's a header 'Price Categories'. Below it, a tab labeled 'GENERAL' is active. The form contains several fields: 'Name' (text input), 'Default Markup' (text input with a value of '0'), 'Cost By' (dropdown menu with 'Carrier' selected), and 'Shared' (checkbox). There are also help icons (question marks) next to 'Default Markup', 'Cost By', and 'Shared'. A hamburger menu icon is visible in the top right corner of the form area.

The data to be configured are the following:

Name, Brief description of the Price Category

Markup, the percentage added by a seller to the cost of a carrier/customer rates to cover expenses and profit in fixing the selling price.

Cost By, defines if the extensions with this category will use the carrier or customer rates. If Customer is selected it is necessary to configure the following options.

Customer Rate, you can add several Customer Rates with different Markup.

Markup, the percentage added by a seller to the cost of a carrier/customer rates to cover expenses and profit in fixing the selling price. If is not defined or is zero the global is taken.

Share, if checked, the price category will be shared with all the tenants.

Dial Codes Categories

This option allows us to obtain more descriptive reports regarding the destination code.

Dial Codes Categories

GENERAL

Name

Description

Save

The data to be configured are the following:

Name, short name of the Dial Code, for example National, International, etc.

Description, brief description of the Dial Code, for example National Calls, International Calls, etc.

Reports

Reports Builder

Report Builder generates all the detailed and summarized reports by extension, cost center, trunks, etc.

Reports Builder

GENERAL

Title ?

Grouped By

Extensions

Type

Detailed

Period

Today

Extensions

☒

Carriers

☒

Default

Cost Centers

☒

Default

Format ?

PDF

Currency

Home Currency (\$)

Call Types

Outgoing Incoming Local Transit

Duration Min/Max ?

0

Infinite

Cost Min/Max

0

Infinite

Hourly Filter

None

Dial Codes Filter

None

Top Limit ?

10

Include Ringing Time ?

☐

Generate Report

Save

The data to be configured are the following:

Title, this represents the title of the report, is required only when the report is saved as template.

Group By, generate the report by grouping it by the selected option.

Type, generate the report with the selected option.

Period, range of date and time to be taken into account in the report.

Extensions, extensions to be included in the report, if the check is selected, all extensions will be included.

Carriers, carriers to be included in the report, if the check is selected, all carriers will be included.

Cost Center, cost center to be included in the report, if the check is selected, all cost centers will be included.

Account Codes, account codes to be included in the report, if the check is selected, all account codes will be included.

Format, defines the format in which the report will be exported.

Currency, currency to show report.

Call Types, types of calls to be included in the report.

Duration Min/Max, duration Min/Max expressed in seconds.

Cost Min/Max, cost Min/Max expressed in home currency.

Hourly Filter, hour filter to use to generate the report. This filter must have been previously created.

Dial Codes Filter, dial codes filter used to generate the report. This filter must have been previously created.

Top Limit, limit the number of items to show in top reports.

You can create a profile for later use in Report Schedule, configure the information and press the “Save” button.

Reports Filters

1. Days and Hours Filter

Days and Hours Filter are used to select only calls that are made within the time set in the filter. This screen shows a filter of working hours.

GENERAL

Filter By

Days and Hours

Description

Days and Hours

Day\Time	00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23
Sunday																								
Monday																								
Tuesday																								
Wednesday																								
Thursday																								
Friday																								
Saturday																								

The data to be configured are the following:

Filter By, type of filter to create, Days and Hours or Prefixes.

Description, brief description of the filter.

Days and Hours, day of the week and hours to select in the filter.

2. Dial Codes Filter

GENERAL

Filter By

Dial Codes

Description

Dial Code

Exclude

Comparison Mode

Call Types

☐

Start With

Add New

The data to be configured are the following:

Filter By, type of filter to create, Days and Hours or Prefixes.

Description, brief description of the filter.

Prefix, prefix to include in the filter.

Exclude, select whether to exclude this prefix when applying this filter.

Comparison Mode, type of comparison to use when comparing the prefix.

Call Types, type of calls in which this prefix will be applied.

Tools

Call Simulator

Simulates the cost of a call, this module is very useful to test the rates.

GENERAL

Source		Carrier	None
Destination		Call Duration	60
Call Direction	Outgoing	Currency	Home Currency (\$)

The data to be configured are the following:

Source, extension that makes the call.

Destination, destination number of the call.

Call Direction, type of call, incoming, outgoing, internal or transit.

Carrier, carrier used to make the call.

Call Duration, call duration.

Currency, allows you to define in which currency the results will be shown. The exchange is made according to the rates' currency.

Users

System Permission

It is possible to limit to which Extensions, Trunks or Cost Centers a user will have access to, for which it is necessary to go to USERS/System Permission.

GENERAL

Description		Cost Centers	Default
Extensions			
Trunks			

- The data to be configured are the following:
- Description, Brief description of System Permission.
 - Extensions, the Extensions that the user will have access to.
 - Trunks, the Trunks that the user will have access to.
 - Cost Center, the Cost Center that the user will have access to.

Settings

Email Settings

To be able to use the module of sending reports automatically it is necessary to configure an email account, to configure this account go to SETTINGS/Email Settings.

GENERAL

Type

Sendmail

From Name

From Email

BCC Email

Test Configuration

Email

The data to be configured are the following:

Type, select the type of email server to use.

SMTP secure, type of security.

Host, email server.

Port, port of the email server.

Username, user to authenticate to the email account.

Password, password to authenticate to the email account.

From Name, name of the owner of the email.

From Email, complete email.

BCC Email, if you want to send a copy of the hidden email, write it here.

Test Configuration, test email settings.



Cloud PBX

INDEX

1.	Introduction.....	6
2.	Initial Configurations	7
2.1	Creating Extensions	7
2.2	Trunks, DIDs, and Outbound Routes.....	7
2.3	Security.....	8
3.	Menu Overview	10
4.	Configuration Considerations	15
4.1	Security	15
4.2	Numbering System.....	15
5.	PBX.....	16
5.1	Extensions	16
5.1.1	Extensions	16
5.1.2	Hot Desking	27
5.1.3	Import Extensions	30
5.1.4	Export Extensions	31
5.1.5	Bulk Modification	31
5.1.6	Extensions Status	32
5.2	Applications	34
5.2.1	Conferences.....	34
5.2.2	Custom Applications.....	37
5.2.3	Custom Destinations	37
5.2.4	Custom Context.....	39
5.2.5	Feature Codes	39
5.2.6	Paging & Intercom	45
5.2.7	Pickup Groups.....	46
5.2.8	Parking	47
5.2.9	Speed Dialing.....	48
5.2.10	Import/Export Speed Dialing	49
5.2.11	Voicemail Broadcast Group	49
5.2.12	Call Back	50
5.2.13	DISA.....	51

5.2.14	PIN List	52
5.2.15	Dynamic Destinations	53
5.3	Class of Service.....	55
5.3.1	Class of Service	55
5.3.2	Feature Categories	56
5.3.3	Dialing Restriction Rules.....	57
5.3.4	Customer Codes.....	59
5.3.5	Authorization Codes.....	59
5.3.6	Route Selections.....	60
5.4	Call Center	61
5.4.1	Ring Groups	61
5.4.2	Queues	63
5.4.3	Queues Priorities	68
5.4.4	Queues VIPs	69
5.4.5	Queues Callback	69
5.5	Calls Routing	72
5.5.1	Trunks.....	72
5.5.2	Outbound Routes.....	76
5.5.3	Inbound Routes (DID)	78
5.5.4	Dynamic Routing (Auto CLIP Routes).....	80
5.6	Incoming Calls Tolls	81
5.6.1	IVR	81
5.6.2	Time Groups	83
5.6.3	Time Conditions.....	84
5.6.4	Announcements	85
5.6.5	Languages	85
5.6.6	Night Mode	86
5.6.7	CID Modifiers	87
5.6.8	CID Lookup.....	88
5.7	Tools	89
5.7.1	PBX CLI	89
5.7.2	Blacklist	89
5.7.3	Dashboard	90
5.7.4	Log File Viewer	91
5.7.5	Cron Profiles	91
5.7.6	Weak Passwords	92
5.8	Extras.....	93
5.8.1	Video Conference	93

5.9	Emergency Calls	94
5.9.1	Emergency Numbers.....	94
5.9.2	Dispatchable Locations.....	95
6.	Reports	96
6.1	CDR Reports.....	96
6.1.1	CDR Filters.....	96
6.1.2	View CDR Reports	97
6.2	PBX Reports.....	98
6.2.1	Active Calls	98
6.2.2	PJSIP Devices	98
6.2.3	SIP Devices.....	99
6.2.4	IAX2 Devices.....	100
6.3	IVR Stats.....	101
6.3.1	IVR Stats.....	101
6.4	Call Center Reports.....	102
6.4.1	Queues Call Back Reports	102
7.	Settings.....	103
7.1	Technology Settings	103
7.1.1	PJSIP Settings.....	103
7.1.2	SIP Settings	104
7.1.3	IAX2 Settings	109
7.1.4	Device Profiles	112
7.1.5	Dial Profiles.....	117
7.2	Voicemail Settings	119
7.2.1	Voicemail Settings.....	119
7.2.2	Voicemail Time Zones	120
7.3	PBX Settings	122
7.3.1	System General	122
7.3.2	AMI	124
7.3.3	Log File	125
7.3.4	RTP Settings	126
7.3.5	CEL Settings.....	127
7.3.6	Mini HTTP Server	127
7.4	Voice Prompts.....	129
7.4.1	PBX Sounds	129
7.4.2	Music on Hold	129

7.4.3	Recording Managements.....	130
8.	Admin.....	131
8.1	Admin.....	131
8.1.1	Users.....	131
8.1.2	Users Profiles	132
8.1.3	Application Keys.....	133
8.1.4	Tenants.....	133
8.2	System Settings.....	137
8.2.1	System Miscellaneous.....	137
8.2.2	Email Settings.....	138
8.2.3	Email Templates.....	140
8.2.4	Certificates.....	140
8.2.5	HTTP Server	141
8.3	Firewall	142
8.3.1	Settings.....	142
8.3.2	Firewall Services	144
8.3.3	Firewall Rules	146
8.3.4	DHCP Settings	146
8.4	Add-ons	150
8.4.1	Add-ons	150
8.5	Tools	153
8.5.1	Backup & Restore	153
8.5.2	Maintenance	153
8.5.3	Branding.....	154
9.	High Availability	156

1. Introduction

Attimo CloudPBX is a Graphic User Interface highly responsive that eases the management of Communications Systems Servers in a quick, intuitive, and safe way.

Attimo CloudPBX provides an intuitive 3-level menu that makes it easy to locate the module you wish to configure.

Some of the usage features from Attimo CloudPBX which you must know about are:

- 1) You can use Attimo CloudPBX from any device and any web browser. Attimo CloudPBX adapts in a transparent way to every device, be it a smartphone, tablet, PC, Mac, or Linux.
- 2) On the top of every screen, you will find a spyglass icon (). When you click on it, you can perform a global search throughout the whole system. Which lets you easily search for extensions, call queues, conferences, DISA, trunks, modules, and much more.
- 3) When navigating through the different modules, you may notice that most of them got a list icon () on the top right-hand corner. When you click on this icon, it will show you the list of all the objects that have been created in that module. On the top part of the list, you will also see a spyglass () that you can use as a filter. Write down any part of the name of the object that you are looking for, and you will see all the objects that coincide with your filter.
- 4) Attimo CloudPBX has been designed so that all the information is in one screen always, without the need to scroll down and lose track of the rest of the configurations. This is why you will see that most of the modules are divided in various tabs that will allow you to see all the information for current object.
- 5) Attimo CloudPBX has a multi-Tab system, which means that you can move to any module you have worked on, from any module. This allows you to work on any module without having to close the prior module you've been working on. There is no need to save all changes when moving between tabs, meaning that you can work on multiple tabs at the same time. For example, if you are creating a Ring Group and need to verify the extensions, you can quickly go to the extensions module and come back to the Ring Group module to finish up.
- 6) The Save and Delete buttons will always be visible on the lower right-hand corner of the screen, no matter the size of the screen you are using.
- 7) You can get more screen real estate by hiding the navigation menu on the left. You can do this by clicking on the () icon on the top left-hand corner of the screen.
- 8) All the fields have a tooltip available if you hover your mouse over their name. In the case of mobile devices, you will have to tap on the text to access the tooltip.
- 9) Obligatory fields are indicated by a communications system next to the field name. For example, Name indicates that the 'Name' field is obligatory.

2. Initial Configurations

Up next, we will explain the steps you can take to have a basic Attimo CloudPBX with some extensions, trunks, and some security measures. For any further explanation of each field and option, the rest of the manual will cover them thoroughly.

2.1 Creating Extensions

To create an extension, you can go to PBX > Extensions > Extensions, and the most important fields are:

Extension, the extension number.

Name, name of the user associated to this extension.

User Device, even though this is filled automatically, you can customize this field and remember it as this is the user, we will use to register our devices.

Password, this field is auto generated, but you can customize it to anything you like. This will be the password used for the device registration.

Always remember that after you create an object, save and afterwards 'Apply Changes' by pressing the red arrows button.



Knowing this, you can create a couple of extensions and we can connect them with any hardware device or softphone, using the following information:

SIP Server, the IP address for our Attimo CloudPBX.

Username, this is the User Device.

Password, the password created on the extension's device.

Once you have the two phones registered you would be able to call between them. Here are a couple of things you can troubleshoot if this is not the case.

The phones are not registering. Verify that the registration port 5060 is enabled on your firewall and that you have entered the correct information.

There is no audio. In this case, it is possible that your server is behind NAT, for which we recommend you find out the Public IP address for your server and if available the Local IP address. With this information, we can go to Settings > Technology Settings > SIP > Networking, and on the NAT section, enter your External address, and on Local Network enter your IP Address. Also, set the NAT option to Force, Comedia. Save and Apply Changes.

2.2 Trunks, DIDs, and Outbound Routes

So, we can communicate with the outside world, it is necessary to connect Attimo CloudPBX with any VoIP trunk provider. There are various providers out there, but we will explain the basic concepts on this example.

Technology, we recommend creating a PJSIP trunk. Almost all serious VoIP providers have this technology type. Though the popular one would be SIP Trunks.

Description, a brief description to identify the trunk.

Class of Service, here we select "Trunk Default". **Local Username**, the username to identify this trunk. **Host**, IP address or hostname for the VoIP Provider.

Remote Username, username provided/used by the VoIP provider to connect with them.

Remote Secret, password used to connect with the VoIP provider.

Contacts, in the case the VoIP provider is providing a PJSIP trunk, here we add the contacts or hosts that they provide. There can be more than one.

From Domain, the domain used on the FROM header.

Qualify, we set this to yes to keep the connection alive.

Contacts, this is a list of IP Addresses or hostnames separated by comma. The IP Addresses can use a subnet mask. The subnet mask can be written in CIDR or Octets separated by period. Separate the IP Address from the netmask with a forward slash "/".

Once we have the trunk connected to our PBX, we can configure our DID so we can receive calls through the trunk we just created. To configure this, we would need to know the DID that we purchased from the Provider, and the format they send it to our Attimo CloudPBX. (For example, 13055605776)

We will now go to PBX > External > Inbound Routes:

Description, a brief description to identify the inbound route.

DID, here we input the DID we have purchased from the VoIP Provider.

Destination, here we can select the destination where we will direct the call to.

Now, we will configure the way we can place outgoing calls from our Attimo CloudPBX. To do this, we go to PBX > External > Outbound Routes, and we configure the following:

Description, a brief description to identify the outbound route.

Trunk, we select the Trunk we wish to use to call outbound. In this case we select the trunk previously created.

Prefix, here we can enter a prefix to select this outbound route whenever we place a call. This prefix will be dropped from the phone number when a call is placed.

Pattern, here we configure the pattern for the numbers we wish to dial. There are a set of variables you can use, and you can see them by hovering over the word 'Pattern'. If you want to use a ten-digit phone number, you can easily enter ten X's, for example XXXXXXXXX.

2.3 Security

It is necessary that we make sure that our Attimo CloudPBX is secure against any possible external attack, for which we make the following recommendations:

Use complex passwords for the extensions you create. (We recommend the auto-generated ones.)

Use complex passwords for the root and admin users.

If you are using a firewall to forward the ports towards your PBX the needed ports are the following:

- **SIP**, ports 5062-5063 UDP/TCP (If you are not going to use the SIP protocol, then there is no need to forward it.)
- **PJSIP**, ports 5060-5061 UDP/TCP If you are not going to use the PJSIP protocol, then there is no need to forward it.)
- **IAX2**, ports 4569 UDP (If you are not going to use the IAX2 protocol, then there is no need to forward it.))
- **RTP**, ports 10000-20000 UDP
- **SSH**, port 22 (This is not necessary if you are not doing remote configurations, so there is no need to publish it.)

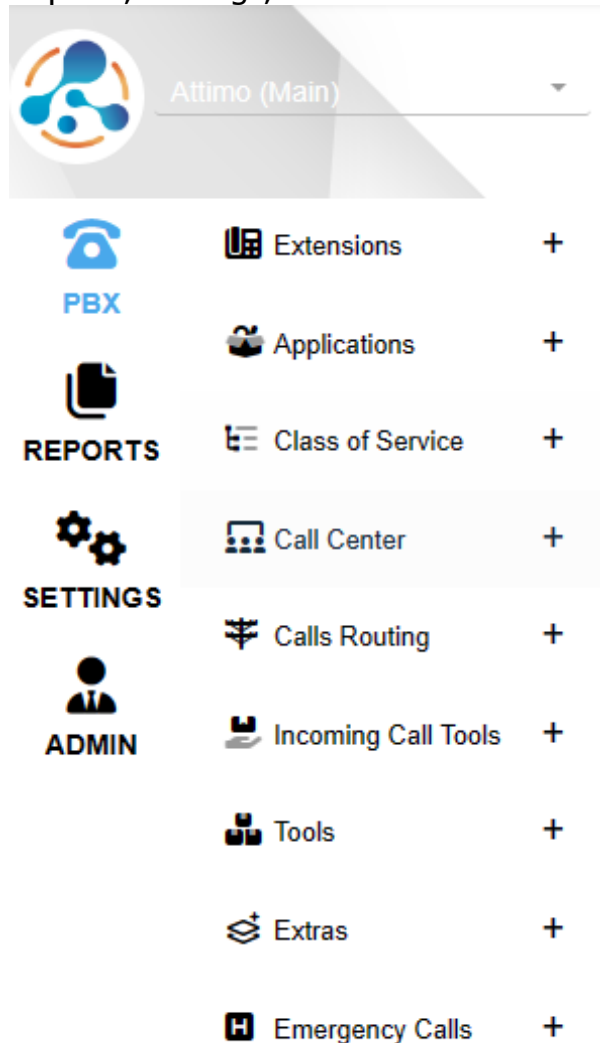
- **HTTP**, port 80 TCP (If you don't need to access the Web UI remotely, there is no need to publish it.)
- **HTTPS**, port 443 TCP, this is only needed if you have a valid certificate on your PBX. (If you don't need to access the Web UI remotely, there is no need to publish it.)
- **Communications Systems HTTP Daemon**, ports 8088-8089 UDP/TCP, only necessary if you are using WebRTC. (If you are not using WebRTC, there is no need to publish it.)
- **VitXi**, ports 6001-6003 TCP, these are necessary for VitXi to work properly. (If you are not using VitXi, there is no need to publish it.)
- **OpenVPN**, port 1194 UDP, if you are using the OpenVPN Server add-on, you will need to forward this port. For better security, you can close every other port on your firewall and only forward this port. You can use the OpenVPN certificates generated by this add-on to access the PBX directly, and many phones nowadays have OpenVPN capabilities.

We recommend the use of the Geo Firewall add-on, which will allow you to limit the access to your PBX in the case it is published. This will prevent people from specific countries to access your PBX.

To increment the security, you can block ICMP requests by going to Admin > Firewall > Settings and disabling the "ICMP requests" option. This will disallow the 'Ping' requests to your PBX.

3. Menu Overview

The Attimo CloudPBX navigation menu is divided into four main sections, PBX, Reports, Settings, and Admin as outlined below.



PBX, where you can find all about the PBX settings:

- Extensions
 - **Extensions**, management of extensions and devices.
 - **Hot Desking**, device management.
 - **Import Extensions**, import extensions from CSV format.
 - **Export Extensions**, export extensions in CSV format.
 - **Bulk Modification**, bulk extension modification.
 - **Bulk Extensions**, create a range of extensions in a single action.
 - **Extension Status**, status of extensions, with possibility to changing the state.
- Applications
 - **Conferences**, conference room management.
 - **Custom Applications**, custom application management.
 - **Custom Destinations**, custom destination management.
 - **Feature Codes**, management of telephone feature codes.

- **Paging & Intercom**, paging & intercom management.
- **Pickup Groups**, management of pickup groups.
- **Parking**, parking management.
- **Speed Dialing**, speed dial management.
- **Import/Export Speed Dialing**, Import/Export speed Dialing from/to CSV format.
- **Voicemail Broadcast Group**, voicemail broadcast group management.
- **Call Back**, management of call back functionality.
- **DISA**, Direct Inward System Access (DISA) management.
- **PIN List**, group of pins that will be used to access outgoing routes.
- **Dynamic Destination**, route calls based on the CID number.
- Class of Service.
 - **Class of Service**, group of settings that define the dial plan to which each extension has access.
 - **Features Category**, telephone feature groups that are associated with a Class of Service.
 - **Dialing Restriction Rules**, dial-up restrictions that are associated with a Class of Service.
 - **Customer Codes**, customer account codes that can be dynamically associated to a call-in order to categorize the call in the CDR.
 - **Authorization Code**, code that authorizes privileges to make a call.
 - **Route Selections**, Automatic Route selection management
- Call Center.
 - **Ring Groups**, groups of extensions that do not handle statistics. Ring Groups allow you to create a single extension number (the Ring Group number) that will ring on multiple extensions.
 - **Queues**, call queues that handle statistics through a log file.
 - **Queues Priorities**, it gives priority to a queue call on another, very useful when an agent serves multiple queues simultaneously.
 - **Queues VIPs**, list of phones that will give priority in the call queue.
 - **Queue Call Back**, with the Queues Call Back module, you can reduce customer frustration by minimizing time spent waiting.
- External
 - **Trunks**, SIP, IAX, DAHDI trunks management.
 - **Outbound Routes**, management prefixes for outgoing routes.
 - **Emergency Numbers**, create groups of emergency numbers to give them priority.
 - **Inbound Routes**, DID management for incoming routes.
 - **Dynamic Routing**, AutoCLIP Routes.

- Incoming Calls
 - **IVR**, IVR and Automatic Attendant management.
 - **Time Groups**, time group management.
 - **Time Conditions**, time conditions management.
 - **Announcement**, pre-announcement management.
 - **Languages**, languages management.
 - **Night Mode**, night mode management.
 - **CID Modifiers**, Modifies the CID in incoming calls.
 - **CID Lookup**, Search caller information in a database or URL.
- Tools
 - **PBX CLI**, Communications Systems command line management interface.
 - **Blacklist**, blacklist management number.
 - **Dashboard**, see system status in real time.
 - **Log File Viewer**, displays the contents of log files.
 - **Cron Profiles**, create profiles for periodic execution of certain routines.
 - **Weak Password**, weak password detection.
 - **Phonebooks**, create Phonebook to be accessed from phones.
- Extras
 - **Video Conference**, create video conferences using WebRTC.
- Emergency Calls
 - **Emergency Number**, this module defines the external numbers that cannot be restricted.
 - **Dispatchable Locations**, this module allows you to create detailed locations which will be used when an Emergency Call is placed.

Reports, where you can find everything about the CDR generated by phone calls.

- CDR Reports
 - **CDR Filters**, management of filters to apply in reports.
 - **CDR**, display call records (CDR).
- PBX Reports
 - **Active Calls**, displays calls in progress in real time.
 - **PJSIP Devices**, shows PJSIP terminals and trunks and if they are registered.
 - **SIP Devices**, shows SIP trunks and devices and if they are registered.
 - **IAX2 Devices**, shows the devices and Trunks.
- **IVR Reports**, create reports of IVR use.

Settings, where you can find everything about the parameters of different technologies (such as SIP and IAX), voice mail settings, the PBX overall event files, configuration of analog and digital interfaces (DAHDI), auto-provisioning of phones (End Point Manager).

- Technology Settings
 - **PJSIP Settings**, general PJSIP settings management.
 - **SIP Settings**, general SIP settings management.
 - **IAX2 Settings**, general IAX settings management.
 - **Profiles**, profile management.
 - **Telephony Settings**, select Tone Zone. It is shown only if

the Telephony module is installed.

- **Dial Profiles**, dial profile management.
- Voicemail Settings
 - **Voicemail Settings**, general voice mail settings management.
 - **Voicemail Timezones**, time zone management for voicemail.
- PBX Settings
 - **System General**, general system settings such as directories, dial-plan settings, etc.
 - **AMI Users**, to create users of AM.
 - **Log File**, create log files related to Communications Systems.
 - **RTP Settings**, general RTP settings management.
 - **CEL Settings**, configure the applications to show in the detailed CDR.
 - **Mini HTTP Server**, Communications Systems provides a basic HTTP/HTTPS server.
- Voice Prompts
 - **PBX Sounds**, manages the voice guides of the system in different languages.
 - **Music on Hold**, to create and upload music on hold.
 - **Recordings Management**, to upload recordings.

Admin, allows you to create system users and manage system settings.

- Admin
 - **Users**, management of system users.
 - **User Profiles**, management of user profiles.
 - **Application Access**, create APIs to give access to third party applications.
- System Settings
 - **System Misc.**, management of system notifications and date and time settings.
 - **Email Settings**, email server configuration.
 - **Certificates**, create certificates of type Self Signed and Let's Encrypt.
 - **HTTP Server**, assign ports to access the interface and enable the HTTPS server.
- Firewall
 - **Settings**, here we configure all the firewall settings as well as intrusion detection (fail2ban).
 - **Services**, we add the services to be applied to the rules later.
 - **Rules**, we apply the firewall rules.
- Network
 - **Network Settings**, network management.
 - **DHCP Settings**, DHCP server configuration.
- Add-ons
 - **Add-ons**, management add-ons modules and software.
- Tools
 - **Backup & Restore**, Backup and Restore the entire PBX configuration.

- **Maintenance,** module to automatically maintain Attimo CloudPBX, convert **recordings**, delete old recordings, etc.
- **Branding,** module to configure Attimo CloudPBX in a personalized way.

4. Configuration Considerations

4.1 Security

Just like any other computer on your network that is connected to the Internet, Attimo CloudPBX can be targeted by hackers for the purpose of making cheap telephone calls. During the entire process of setting up Attimo CloudPBX, you should be constantly aware of the potential security implications of each step and make sure that your system is well protected.

4.2 Numbering System

You need to decide how many digits to use for extensions – do you want to use 3, 4, or more? You should consider that most feature codes are 2 digits, so setting a system with 2-digit extensions is not practical. It will help you to navigate your system if you group similar functions together, for example, by using the following ranges:

2000 - 2999 for extensions

500 - 599 for ring groups

600 - 699 for queues

700 - 799 for conferences

5. PBX

5.1 Extensions

5.1.1 Extensions

Extensions allow you to configure extensions (users) and devices (telephones) in your system.

General

[GENERAL](#)[VOICEMAIL](#)[RECORDING](#)[ADVANCED](#)[FOLLOW ME](#)[INCOMING ROUTES](#)

Extension *

1002

Name *

Class of Service

All Permissions

Features Password

*67676

Email Addresses

Internal CID

Name

Number

External CID

Name

Number

Emergency CID

Name

Number

Account Code

Language

Portuguese (Brazil) (pt_BR)

Devices

Technology

PJSIP

IAX2

VIRTUAL

NONE

User Device *

1002

Password *

Device Description *

1002

Profile

Default PJSIP Profile

Max Contacts

5

Codecs

DTMF Mode

rfc4733

Emergency CID

Name

Number

Dispatchable Location

Default

Deny

0.0.0.0

Permit

0.0.0.0

Ring Device

Yes

WebRTC Client

No

Save

Extension, number to dial to reach this extension. The extension number must be unique, and should not conflict with an existing extension number, or any other number that is assigned to any other entity within the system, such as a conference, queue, ring group, feature code, etc. The value of this field cannot be changed after the extension has been saved.

Name, name to identify this extension. This is generally the end user's name or the location of the extension, e.g. Fernando Alonso or Server Room. This value will be displayed as the caller ID text for any calls placed from this extension to other users or devices on the PBX unless the Internal CID field contains a value.

CoS Name, the dial plan can be segmented into sections, called Classes of Service (CoS). CoS are the basic organizational unit within the dial plan, and as such, they keep different sections of the dial plan independent of each other. Attimo CloudPBX uses CoS to enforce security boundaries between the various parts of the dial plan, as well as to provide different classes of service to different groups of users.

Features Password, password to access certain system features and the control panel of the phone.

Email Address, email address to where the services messages will be sent.

Internal CID, internal Caller ID for the extension, consisting of two parts: the CID Name and the CID Number. This will define the caller ID text that is displayed when

this user calls other (internal) users on the same PBX. This could be used when a user is part of a department in which callbacks should be directed to the department rather than directly to the user (such as a technical support department). This field is not mandatory. If the field is left blank, the user's extension will be used to set the Outbound Caller ID text.

External CID, external Caller ID for the extension, consisting of two parts: the CID Name and the CID Number. This will define the caller ID text that is displayed when this user makes calls outside of the PBX. This could be used when a user is part of a department in which callbacks should be directed to the department rather than directly to the user (such as a technical support department). Setting the caller ID must be supported by the trunk service provider. This field is not mandatory, but if the field is left blank, the default caller ID name for the trunk placing the call will be used to set the caller ID name text.

Emergency CID, it allows to define the caller id that will be used in case of calling an emergency number.

Account Code, this field is used to populate the Account Code field of the Call Detail Record (CDR). If the field is left blank, the Account Code field of the CDR record will also be blank.

Language, specifies the language setting to be used for this extension. This will force all prompts specific to the user to be played in the selected language, provided that the language is installed, and voice prompts for the specified language exist on your server. This field is not required. If left blank, prompts will be played in the default language of the Attimo CloudPBX server.

Devices section

This section allows you to configure the device that is linked to the extension. Technology, type of technology used by this device. The technology options are:

PJSIP, PJSIP device **SIP**, SIP device **IAX2**, IAX device **FXS**, analog device **NONE**, extension without device.

PJSIP

User Device, username to be used when registering this device.

Password, password (secret) associated with this device. Passwords can be the weakest link on any externally accessible PBX system, as malicious users will attempt to locate extensions having weak passwords. Extensions that authenticate by using simple passwords such as "1234" stand a good chance of being compromised, allowing an attacker to place calls through your PBX. Pick strong passwords carefully and ensure that passwords are not given to anyone who does not need to know them. Passwords should be at least 8 characters long and should include a random mixture of letters (both upper- and lower-case), numbers, and special characters.

Profile, group of settings for this device. Each technology (PJSIP, SIP, IAX2, DAHDi, or None) must have at least one (default) profile that defines attributes for the technology. You can configure these profiles in the Settings->Technology Settings->Profiles menu.

Max Contacts, maximum number of contacts that can bind to an AoR.

Codecs, list of allowed codecs. The order in which the codecs are listed determines their order of preference. If you select at least one codec, the

DISALLOW=ALL parameter will be added. This will ensure that the device will only use only the codecs that you specifically define for the device.

DTMF Mode, sets default dtmf-mode for sending Dual Tone Multi-Frequency (DTMF). The DTMF mode for a SIP device specifies how touchtone will be transmitted to the other side of the call. The default value is rfc4733. Available options are:

- Rfc4733.
- info: SIP INFO messages (application/dtmf-relay).
- shortinfo: SIP INFO messages (application/dtmf).
- inband: Inband audio (requires 64 kbit codec -alaw, ulaw).
- auto: Use rfc4733 if offered, in-band otherwise.

Device Description, a short (optional) description to identify this device.

Emergency CID, this is the CID to be used when an Emergency Call is placed with this device.

Dispatchable Location, this is the location to be used whenever an Emergency Call is placed from this device.

Deny, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. This option should be in the format of an IP address and subnet, such as 192.168.25.10/255.255.255.255 (denies traffic from this specific IP address), or 192.168.1.0/255.255.255.0 (to disallow traffic for this extension from the IP range of 192.168.1.1 to 192.168.1.254). It is possible to enter a value of 0.0.0.0/0.0.0.0 to deny all the networks by default, and, to enter specific networks from which traffic can be accepted in the permit option. This option is commonly used to restrict endpoint usage to a particular network, so that if the endpoint is stolen or otherwise removed from the network, it cannot be used to place calls and will be essentially useless. This field is not required. If it is left blank, Attimo CloudPBX will not block traffic for this peer from any IP address.

Permit, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. For example, 192.168.10.0/255.255.255.0 allows traffic from any address on the 192.168.10.x network. The permit option is the opposite of the deny option. Specific IP addresses or networks can be added in this option to allow traffic for this extension from the entered IP/network. This field is not required. If it is left blank, traffic will be allowed from all IP addresses. Strengthen your system security by use of the deny and allow options, where possible. If the endpoint is static, we strongly recommend that you make proper use of the permit and deny options to ensure that traffic is only allowed from the specific address. Even if the endpoint is not static, but always resides on a known subnet, you should limit the allowed range to that specific subnet.

Ring Device, this determines whether incoming calls should cause the device to ring. **Send Push**, if this is enabled, the PBX will attempt to send push notifications to mobile devices using **VitXi Mobile**.

VitXi Client, if this is enabled, this device can be used for VitXi Applications, be it WebRTC or Mobile Apps. When devices with this option enabled are use on VitXi Mobile, all the premium features will be enabled on the softphone.

SIP

User Device*, username to be used when registering this device.

Password, password (secret) associated with this device. Passwords can be the weakest link on any externally accessible PBX system, as malicious users will attempt to locate extensions having weak passwords. Extensions that authenticate by using simple passwords such as "1234" stand a good chance of being compromised, allowing an attacker to place calls through your PBX. Pick strong passwords carefully and ensure that passwords are not given to anyone who does not need to know them. Passwords should be at least 8 characters long and should include a random mixture of letters (both upper- and lower-case), numbers, and special characters.

Profile, group of settings for this device. Each technology (SIP, IAX2, DAHDi) must have at least one (default) profile that defines attributes for the technology. You can configure these profiles in the Settings->Technology Settings->Profiles menu.

Codecs, list of allowed codecs. The order in which the codecs are listed determines their order of preference. If you select at least one codec, the DISALLOW=ALL parameter will be added. This will ensure that the device will only use only the codecs that you specifically define for the device.

NAT, (Network Address Translation) is a technology commonly used by firewalls and routers to allow multiple devices on a LAN with 'private' IP addresses to share a single public IP address. A private IP address is an address, which can only be addressed from within the LAN, but not from the Internet outside the LAN Options:

- **Default:** will use the default NAT settings set on the SIP Settings.
- **No:** No special NAT handling other than RFC3581.
- **Force:** Pretend there was a report parameter even if there wasn't.
- **Comedia:** Send media to the report Communications Systems received it from regardless of where the SDP says to send it.
- **Auto Force:** Set the force report option if Communications Systems detects NAT.
- **Auto Comedia:** Set the comedia option if Communications

Systems detects NAT. **DTMF Mode**, sets default dtmf-mode for sending Dual Tone Multi-Frequency (DTMF). The DTMF mode for a SIP device specifies how touchtone will be transmitted to the other side of the call. The default value is rfc2833. Available options are:

- **info:** SIP INFO messages (application/dtmf-relay).
- **shortinfo:** SIP INFO messages (application/dtmf).
- **inband:** Inband audio (requires 64 kbit codec -alaw, ulaw).
- **auto:** Use rfc2833 if offered, in-band otherwise.

Device Description, a short (optional) description to identify this device.

Deny, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. This option should be in the format of an IP address and subnet, such as 192.168.25.10/255.255.255.255 (denies traffic from this specific IP address), or 192.168.1.0/255.255.255.0 (to disallow traffic for this extension from the IP range of 192.168.1.1 to 192.168.1.254). It is possible to enter a value of 0.0.0.0/0.0.0.0 to deny all the networks by default, and, to enter specific networks from which traffic can be accepted in the permit option. This option is commonly used to restrict endpoint usage to a particular network, so that if the endpoint is stolen or otherwise removed from the network, it cannot be used to place calls and will be essentially useless. This field is not required. If it is left blank, Attimo CloudPBX will not block traffic for

this peer from any IP address.

Permit, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. For example, 192.168.10.0/255.255.255.0 allows traffic from any address on the 192.168.10.x network. The permit option is the opposite of the deny option. Specific IP addresses or networks can be added in this option to allow traffic for this extension from the entered IP/network. This field is not required. If it is left blank, traffic will be allowed from all IP addresses. Strengthen your system security by use of the deny and allow options, where possible. If the endpoint is static, we strongly recommend that you make proper use of the permit and deny options to ensure that traffic is only allowed from the specific address. Even if the endpoint is not static, but always resides on a known subnet, you should limit the allowed range to that specific subnet.

Ring Device, this determines whether incoming calls should cause the device to ring.

IAX2

User Device*, username to be used when registering this device.

Password, password (secret) associated with this device. Passwords can be the weakest link on any externally accessible PBX system, as malicious users will attempt to locate extensions having weak passwords. Extensions that authenticate by using simple passwords such as "1234" stand a good chance of being compromised, allowing an attacker to place calls through your PBX. Pick strong passwords carefully and ensure that passwords are not given to anyone who does not need to know them. Passwords should be at least 8 characters long and should include a random mixture of letters (both upper- and lower-case), numbers, and special characters.

Profile, group of settings for this device. Each technology (SIP, IAX2, Telephony, or None) must have at least one (default) profile that defines attributes for the technology. You can configure these profiles in the Settings->Technology Settings->Profiles menu.

Codecs, list of allowed codecs. The order in which the codecs are listed determines their order of preference. If you select at least one codec, the DISALLOW=ALL parameter will be added. This will ensure that the device will only use only the codecs that you specifically define for the device.

Device Description, a short (optional) description to identify this device.

Deny, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. This option should be in the format of an IP address and subnet, such as 192.168.25.10/255.255.255.255 (denies traffic from this specific IP address), or 192.168.1.0/255.255.255.0 (to disallow traffic for this extension from the IP range of 192.168.1.1 to 192.168.1.254). It is possible to enter a value of 0.0.0.0/0.0.0.0 to deny all of the networks by default, and, to enter specific networks from which traffic can be accepted in the permit option. This option is commonly used to restrict endpoint usage to a particular network, so that if the endpoint is stolen or otherwise removed from the network, it cannot be used to place calls and will be essentially useless. This field is not required. If it is left blank, Attimo CloudPBX will not block traffic for this peer from any IP address.

Permit, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. For example, 192.168.10.0/255.255.255.0 allows traffic from any address on the 192.168.10.x network. The permit option is the opposite of the deny option. Specific IP addresses or networks can be added in this option to allow

traffic for this extension from the entered IP/network. This field is not required. If it is left blank, traffic will be allowed from all IP addresses. Strengthen your system security by use of the deny and allow options, where possible. If the endpoint is static, we strongly recommend that you make proper use of the permit and deny options to ensure that traffic is only allowed from the specific address. Even if the endpoint is not static, but always resides on a known subnet, you should limit the allowed range to that specific subnet.

Ring Device, this determines whether incoming calls should cause the device to ring.

FXS – (Only available if the DAHDI add-on is installed)

Channel*, the Telephony (DAHDI) channel, selected from the drop-down list, that should be associated with this device.

Profile, group of settings for this device. Each technology (SIP, IAX2, Telephony, or None) must have at least one (default) profile that defines attributes for the technology. You can configure these profiles in the Settings->Technology Settings->Profiles menu.

Device Description, a short (optional) description to identify this device.

Ring Device, this determines whether incoming calls should cause the device to ring.

NONE

- Extensions that do not have a device, used for virtual voicemail or Hot Desking.

Voicemail

GENERAL VOICEMAIL RECORDING ADVANCED FOLLOW ME INCOMING ROUTES

Enabled	☒	Allow to Call Back	☐
Voicemail Password	1002	Ask Password	☒
Zone Messages	None	Skip Instructions	☐
Alias		Say CID	☒
Busy Prompt	None	Say Duration	☒
Unavailable Prompt	None	Envelope	☒
Attach Voicemail	☒	Hide From Directory	☐
Delete	☐	Allow to Dial Out	☐
		Generate Hint	☐

Operator Destination

Enabled, enable or disable voicemail. If voicemail is not enabled, voicemail messages cannot be left for the user.

Attach Voicemail, Attach voicemail to email.

Delete, the voicemail is deleted from the server after the voicemail has been delivered. Be careful with this option, because Attimo CloudPBX will allow you to delete the message without guaranteeing that a copy of it has been attached to the email notification, or that the email has been delivered successfully. This could mean that after a message is left and a notification email is sent to the user, the actual voicemail that was left may no longer be accessible.

Voicemail Password, the numeric password to access the voicemail. The voicemail system will compare the password entered by the user against this value.

Zone Messages, time zone for messages. If not set, the time zone will be taken from the general settings section. Irrelevant if envelope is no.

Alias, an alternative name that can be used in the system-created phonebook, or for dialing using the Phonebook Directory feature code (411).

Allow to Call Back, if checked, users will be available to call back to the sender of a message. The specified Class of Service will need to be able to handle dialing of numbers in the format in which they are received (for example, the country code may not be received with the caller ID but might be required for the outgoing call).

Ask Password, it allows to define if the users who dials *97 to access to their own voicemail will be prompted to enter its voicemail password or not. This doesn't apply for the "Remote Voicemail (*98)" feature.

Skip Instructions, if set to yes, it will skip the playback of instructions for leaving a message to the calling party.

Say CID, system will play back the caller ID number of the person who left the message prior to the message being played.

Say Duration, turn on/off the duration information before playing the voicemail message.

Envelope, this determines whether the user will hear the date and time that the message was left prior to hearing the voicemail message being played.

Hide from Directory, hide If set to yes, this name of this user will not be visible to the system-created phonebook, and you cannot dial to this user using the Phonebook Directory feature code (411).

Allow to Dial Out, if allowed, users can dial out from their mailboxes (option 4 from mailbox's advanced menu). **This is considered a very dangerous practice in a phone system (mainly because many voicemail users like to use 1234 as their password) and is therefore not recommended.**

Generate Hint, if enabled, it will be possible to remotely monitor the voicemail status of this extension through a BLF key. To configure the BLF, you must use the following format: vm_1234, where 1234 is the extension that will be monitored.

Recording

The screenshot shows the 'Recording' configuration tab. At the top, there are navigation links: GENERAL, VOICEMAIL, **RECORDING**, ADVANCED, FOLLOW ME, and INCOMING ROUTES. The main content area is divided into two columns. The left column contains 'Outgoing' (No), 'Incoming' (No), 'Dictation' (Enabled: No, Format: Ogg Vorbis), and 'Auto-Send Email' (No). The right column contains 'Internal' (No), 'On Demand Recording' (No), and 'Auto-Send Email' (No). A large grey rectangular area is visible below the configuration fields. A 'Save' button is located at the bottom right.

In this tab you will find the information about the recording telephone calls and dictation recording.

This group of fields allows a user to control the recording of incoming or outgoing calls. The user can either dial a feature code (*3) to selectively enable recording for the current call, never record calls, or always record calls.

Outgoing, record external outgoing calls. **Incoming**, record external incoming calls. **Internal**, record internal calls.

On Demand Recording, record calls on demand.

Dictation section

- **Enabled**, activates the dictation service when set to Yes.
- **Format**, recording audio format:
 - OGG Vorbis.
 - GSM.
 - WAV.

Auto-Send Email, recording will be sent automatically once completed.

Advanced

The screenshot shows the 'Advanced' configuration tab. At the top, there are navigation links: GENERAL, VOICEMAIL, RECORDING, **ADVANCED**, FOLLOW ME, and INCOMING ROUTES. The main content area is divided into two columns. The left column contains 'Ring Time' (Default (30)), 'Call Limit' (No Limit), 'Internal Auto Answer' (Disable), 'Dial Profile' (Default), 'Music on Hold Class' (Default), 'Secretary Extension' (None), 'CallerID On Diversions' (Callee), and 'Notify Missed Calls'. The right column contains 'Diversion Hints' (No), 'Block Spy Me' (No), 'Send CallerID' (Yes), 'Call Waiting' (Yes), 'Pinless' (No), 'Dynamic Routing' (No), and 'Dynamic External CID' (No). Below these fields, there is a 'Call Center Settings' section with 'Dynamic Queues' and 'Static Queues' (both empty). A 'User Portal' section contains 'Enable Portal' (No), 'Portal User' (1002), and 'Portal Password'. A 'User Image' section contains a placeholder image and a 'Select Image' button. A 'Save' button is located at the bottom right.

Ring Time, the number of seconds to ring the device before giving up and moving on to the next priority for the extension.

Call Limit, maximum number of simultaneous calls that can be received by this device.

Dial Profile, there are many options that you can set on the outbound call, including call screening, distinctive ringing, and more. Goto Settings/Technology/Dial Profile for more information.

Internal Auto Answer, automatic call answering can be requested from within the incoming call by using the SIP Alert-Info header. This can only be utilized when automatic call answering is allowed on the phone.

Music on Hold Class, this option specifies which music on hold class to suggest to the peer channel when this channel places the peer on hold.

Secretary Extension, functionality is used to re-route all incoming calls for this extension to the secretary's extension. Only the secretary is allowed to make direct calls to this extension.

Caller ID on Diversions, this allows you to define which CID will be sent when the call is forwarded.

- **Caller**, this will send the CID of the person calling.
- **Called**, this will send the CID of the person receiving the call and has the diversion activated.

Diversions Hints, this generates hints regarding status of the extension. For example, hints could be generated for diversions (DND, Call Forwarding, Personal Assistant and Boss/Secretary). **Do not activate this option unless your phone has a console or keys for Hints. Activating this option can slow down the "Apply Changes" in the PBX and overload.**

Block Spy Me, do not let other users to spy on this extension.

Send Caller ID, send, or hide, the Caller ID for this extension.

Call Waiting, if you uncheck this option, only one incoming call will be allowed to this extension.

Pin less, if enabled, the user of this extension will not be prompted to enter pin on outbound routes that have assigned a pin set.

Dynamic Routing, this allows you to enable or disable the dynamic routing towards this extension. If enabled, when an external party (that was previously called by this extension) calls back, the call will be routed directly to this extension.

Call Center Settings

This section contains two fields (Dynamic Queues, Static Queues) that allows you to assign or remove massively an extension to any queue or group of queues.

Dynamic Queues, these are the agents who will be allowed to log in the call queue. **Static Queues**, are agents that will always be in the queue, these agents do not need to log in.

User Portal

Enable Portal, allow users to login to Portal to configure their own extension.

Portal User, user for login as portal user.

Portal Password, password for access to portal area.

User Image section

Allows the user to select any image and associate it with the extension. It may be the photo of the owner of the extension, an avatar, or any other graphic, in PNG, JPG, or JPEG format. The size of the file must be less than 20 MB.

Follow Me

GENERAL VOICEMAIL RECORDING ADVANCED **FOLLOW ME** INCOMING ROUTES

Follow Me List

Initial Ring Time 15

Ring Time Default (30)

Ring Strategy One by One

Music on Hold Class Default

Call-from prompt Default

No Recording Prompt Default

Please Hold Prompt None

Status Prompt None

Sorry Prompt None

Enabled No

FollowMe Options

Record Caller's Name No

Prompt Callee No

Save

Follow Me List, list of extensions and/or external numbers to be accessed by follow me.

Initial Ring Time, time in seconds to ring the primary extension before calling to the members on the follow-me list.

Ring Time, this is the time that the phone will be allowed to ring, without being answered, before continuing to an alternative destination.

Ring Strategy, here you define the strategy to ring this list.

One by One: ring all available number in the Follow List One by One.

Ring All: ring all available number in the Follow List at the same time.

Music on Hold, the Music on Hold class that should be used for the caller while they are waiting to be connected.

Call-from Prompt, you can select the default option to use the "Incoming call from" message prompt or use your own custom prompt.

No Recording Prompt, you can select to use the standard "You have an incoming call" message prompt when the caller elects not to leave their name, or the option isn't set for them to do so or use your own custom prompt.

Please Hold Prompt, you can select to use the standard "Please hold while we try and connect your call" message prompt or use your own custom prompt.

Status Prompt, you can select to use the standard "The party you're calling isn't at their desk" message prompt or use your own custom prompt.

Sorry Prompt, you can select to use the standard "I'm sorry, but we were unable to locate your party" message prompt or use your own custom prompt.

Enabled, it allows you to enable/disable the follow-me feature on this extension.

Follow Me Options

Record Caller's Name, here you can record the caller's name so it can be announced to the callee at each step.

Prompt Called, called party will be asked whether they wish to accept the incoming call.

Incoming Routes

GENERAL	VOICEMAIL	RECORDING	ADVANCED	FOLLOW ME	INCOMING ROUTES	
Description		CID Pattern				
DID Pattern						

The DID number for incoming calls, i.e. the inbound route that should be associated with this extension.

Description, a short description to identify the route.

DID Pattern, the DID number for incoming calls, i.e. the inbound route that should be associated with this extension.

CID Pattern, optional CID number to make route more specific.

Actions, go to Inbound Route module.

Welcome Email

Starting from Version 3 of Attimo CloudPBX, it is now possible to send a welcome email with various information about the new user extension. For this to work, we need to have the following configured.

Under Settings > PBX Settings > System General, enable the option "Send Welcome Email".

Configure the email client with the account to send emails with, under Admin > System Settings > E-Mail Settings.

Make sure that the extension you created has an email assigned on the General Tab.
Note: If you wish to modify the contents of this email, you can do so under Admin > System Settings > Email Templates.

You can also see the QR code or send the Welcome Email Manually by pressing the QR button next to the Device Field.

5.1.2 Hot Desking

The Hot Desking module is where the accounts are created for devices without the need of having an extension number. A Hot Desking device is associated with an extension that previously had to be created in the module extensions with technology option "None", i.e. without being associated with any device. A Hot Desking device can be associated with an extension by dialing the hot desking feature code (*80), the extension number, and the extension password. To remove the association, you only need to dial the hot desking feature code (*80).

General

The screenshot shows the 'General' configuration page for a Hot Desking device. The page is divided into two columns. The left column contains fields for Technology (PJSIP, IAX2), User Device *, Password *, Device Description *, Profile (Default PJSIP Profile), Max Contacts, Codecs, and DTMF Mode (rfc4733). The right column contains fields for Emergency CID (Name, Number), Dispatchable Location (Default), Deny (x0.0.0.0/0), Permit (x0.0.0.0/0), Ring Device (Yes), and WebRTC Client (No).

Technology, type of technology for this device. There are four options:

- **PJSIP**, PJSIP device.
- **SIP**, sip device.
- **IAX2**, iax device.
- **FXS**, analog/digital device. This is displayed if you got the DAHDI module installed.

Emergency CID, this is the Emergency CID to use when placing an Emergency Call from this device.

Dispatchable Location, Location to use whenever an Emergency Call is places.

PJSIP

User Device*, username to be used when registering this device.

Password, password (secret) associated with this device. Passwords can be the weakest link on any externally accessible PBX system, as malicious users will attempt to locate extensions having weak passwords. Extensions that authenticate by using simple passwords such as "1234" stand a good chance of being compromised, allowing an attacker to place calls through your PBX. Pick strong passwords carefully and ensure that passwords are not given to anyone who does not need to know them. Passwords should be at least 8 characters long and should include a random mixture of letters (both upper- and lower-case), numbers, and special characters.

Profile, group of settings for this device. Each technology (PJSIP, SIP, IAX2, DAHDI, or None) _ have at least one (default) profile that defines attributes for the technology. You can configure these profiles in the Settings->Technology Settings->Profiles menu.

Max Contacts, maximum number of contacts that can bind to an AoR.

Codecs, list of allowed codecs. The order in which the codecs are listed determines their order of preference. If you select at least one codec, the `DISALLOW=ALL` parameter will be added. This will ensure that the device will only use only the codecs that you specifically define for the device.

DTMF Mode, sets default dtmf-mode for sending Dual Tone Multi-Frequency (DTMF). The DTMF mode for a SIP device specifies how touchtone will be transmitted to the other side of the call. The default value is `rfc4733`. Available options are:

- `Rfc4733`.
- `info`: SIP INFO messages (`application/dtmf-relay`).
- `shortinfo`: SIP INFO messages (`application/dtmf`).
- `inband`: Inband audio (requires 64 kbit codec `-alaw`, `ulaw`).
- `auto`: Use `rfc4733` if offered, in-band otherwise.

Device Description, a short (optional) description to identify this device.

Deny, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. This option should be in the format of an IP address and subnet, such as `192.168.25.10/255.255.255.255` (denies traffic from this specific IP address), or `192.168.1.0/255.255.255.0` (to disallow traffic for this extension from the IP range of `192.168.1.1` to `192.168.1.254`). It is possible to enter a value of `0.0.0.0/0.0.0.0` to deny all the networks by default, and, to enter specific networks from which traffic can be accepted in the permit option. This option is commonly used to restrict endpoint usage to a particular network, so that if the endpoint is stolen or otherwise removed from the network, it cannot be used to place calls and will be essentially useless. This field is not required. If it is left blank, Attimo CloudPBX will not block traffic for this peer from any IP address.

Permit, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. For example, `192.168.10.0/255.255.255.0` allows traffic from any address on the `192.168.10.x` network. The permit option is the opposite of the deny option. Specific IP addresses or networks can be added in this option to allow traffic for this extension from the entered IP/network. This field is not required. If it is left blank, traffic will be allowed from all IP addresses. Strengthen your system security by use of the deny and allow options, where possible. If the endpoint is static, we strongly recommend that you make proper use of the permit and deny options to ensure that traffic is only allowed from the specific address. Even if the endpoint is not static, but always resides on a known subnet, you should limit the allowed range to that specific subnet.

Ring Device, this determines whether incoming calls should cause the device to ring.

SIP

User*, user to register this device.

Password, password (secret) associated with this device.

Profile, technology profile to be associated with this device.

Codecs, list of allowed codecs. The order in which the codecs are listed determines their order of preference. If you do not select at least one codec, `DISALLOW=ALL` will be used.

DTMF Mode, sets default dtmf-mode for sending Dual Tone Multi-Frequency (DTMF). The DTMF mode for a SIP device specifies how touchstones will be transmitted to the other side of the call. The default value is `rfc2833`. Available

options are:

- info: SIP INFO messages (application/dtmf-relay).
- shortinfo: SIP INFO messages (application/dtmf).
- inband: Inband audio (requires 64 kbit codec -alaw, ulaw).
- auto: Use rfc2833 if offered, in-band otherwise.

Device Description*, a short description to identify this device.

Deny, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. This option should be in the format of an IP address and subnet, such as 192.168.25.10/255.255.255.255 (denies traffic from this specific IP address), or 192.168.1.0/255.255.255.0 (to disallow traffic for this extension from the IP range of 192.168.1.1 to 192.168.1.254). It is possible to enter a value of 0.0.0.0/0.0.0.0 to deny all of the networks by default, and, to enter specific networks from which traffic can be accepted in the permit option. This option is commonly used to restrict endpoint usage to a particular network, so that if the endpoint is stolen or otherwise removed from the network, it cannot be used to place calls and will be essentially useless. This field is not required. If it is left blank, Attimo CloudPBX will not block traffic for this peer from any IP address.

Permit, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. For example, 192.168.10.0/255.255.255.0 allows traffic from any address on the 192.168.10.x network. The permit option is the opposite of the deny option. Specific IP addresses or networks can be added in this option to allow traffic for this extension from the entered IP/network. This field is not required. If it is left blank, traffic will be allowed from all IP addresses. Strengthen your system security by use of the deny and allow options, where possible. If the endpoint is static, we strongly recommend that you make proper use of the permit and deny options to ensure that traffic is only allowed from the specific address. Even if the endpoint is not static, but always resides on a known subnet, you should limit the allowed range to that specific subnet.

NAT, (Network Address Translation) is a technology commonly used by firewalls and routers to allow multiple devices on a LAN with 'private' IP addresses to share a single public IP address. A private IP address is an address, which can only be addressed from within the LAN, but not from the Internet outside the LAN Options:

- No: No special NAT handling other than RFC3581.
- Force: Pretend there was a report parameter even if there wasn't.
- Comedia: Send media to the port Communications Systems received it from regardless of where the SDP says to send it.
- Auto Force: Set the force_report option if Communications Systems detects NAT.
- Auto Comedia: Set the comedia option if Communications Systems detects NAT.

Ring Device, this determines whether incoming calls should cause this device to ring.

IAX2

User*, user to register this device.

Password, password (secret) associated with this device.

Profile, technology profile to be associated with this device.

Codecs, list of allowed codecs. The order in which the codecs are listed determines their order of preference. If you do not select at least one codec, DISALLOW=ALL will be used.

Device Description*, a short description to identify this device.

Deny, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. This option should be in the format of an IP address and subnet, such as 192.168.25.10/255.255.255.255 (denies traffic from this specific IP address), or 192.168.1.0/255.255.255.0 (to disallow traffic for this extension from the IP range of 192.168.1.1 to 192.168.1.254). It is possible to enter a value of 0.0.0.0/0.0.0.0 to deny all of the networks by default, and, to enter specific networks from which traffic can be accepted in the permit option. This option is commonly used to restrict endpoint usage to a particular network, so that if the endpoint is stolen or otherwise removed from the network, it cannot be used to place calls and will be essentially useless. This field is not required. If it is left blank, Attimo CloudPBX will not block traffic for this peer from any IP address.

Permit, in a user/peer definition, allows you to limit SIP traffic to and from this peer to a certain IP or network. For example, 192.168.10.0/255.255.255.0 allows traffic from any address on the 192.168.10.x network. The permit option is the opposite of the deny option. Specific IP addresses or networks can be added in this option to allow traffic for this extension from the entered IP/network. This field is not required. If it is left blank, traffic will be allowed from all IP addresses. Strengthen your system security by use of the deny and allow options, where possible. If the endpoint is static, we strongly recommend that you make proper use of the permit and deny options to ensure that traffic is only allowed from the specific address. Even if the endpoint is not static, but always resides on a known subnet, you should limit the allowed range to that specific subnet.

Ring Device, this determines whether incoming calls should cause this device to ring.

FXS – (Only appears when the DAHDI add-on is installed.)

Channel*, the Telephony (DAHDI) channels, selected from the drop-down list, to be associated with this device.

Profile, technology profile to be associated with this device.

Device Description*, a short description to identify this device.

Ring Device, this determines whether incoming calls should cause this device to ring.

5.1.3 Import Extensions

Import Extensions is an easy way to create extensions in a large system. You can create a csv file from a template that can be downloaded from this same module. This template can be edited in Excel and then imported into Attimo CloudPBX.

General

GENERAL

CSV File

[Import Extensions](#) [Download Import Format](#)

CSV File, CSV File with details of the extension/s to process.

An example of the file format can be download by press the "Download Import Format" button at the bottom of the screen. In the first line of this file there is a complete description of each field.

5.1.4 Export Extensions

Export all Extensions in CSV format.

GENERAL

Export Extensions

CSV Format

	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P
1	Operation mode	Extension number	Display name	Language selected	Class of Service	Device technology	Profile for device	Device user	Device password	Device description	Caller ID name	Caller ID number	FXS channel	Ring device	Password for email	Email address
2		extension	ext_name	language	class_of_service	technology	profile_name	device_user	device_password	device_description	devices_email	devices_email	channel	ring_device	features_password	email
3	add	2000	Boss	en	all	pjsip	Default PJSIP	2000	hn7Qx8Gbbn	Boss				yes	*26382	
4	add	2001	Secretary	en	all	sip	Default SIP P	2001	pSWjHjS5B2	Secretary				yes	*88832	
5	add	2002	IT Admin	en	all	iax	Default IAX2	2002	WuQVYbI5aj	IT Admin				yes	*93775	

5.1.5 Bulk Modification

In this module, you can make changes to a group of extensions very easily and quickly. For example, you could change the language of all the extensions at once.

General

GENERAL

Extensions List *

1000 - TESTE ,1001 - 1001

Field *

Select a Field

Click on the Add Extensions button to select the extensions that you wish to modify.

Field, manage the following fields:

Class of Service Ring Time Language Account Code Dial Options Music on Hold Call Recordings Diversion Hints

5.1.6 Extensions Status

This module shows the status of all extensions with the option to change any status by simply pressing the (✎) button that is located at the end of each line.

General

GENERAL

Show 100 entries

Search:

Extension	Boss/Secretary	Secretary Extension	Personal Assistant	Follow-me	DND	CFI	CFB	CFN	CFU	Call Completion	Devices	Actions
1000 - TESTE	☐		☐	☐	☐	☐	☐	☐	☐	☐		
1001 - 1001	☐		☐	☐	☐	☐	☐	☐	☐	☐		

Showing 1 to 2 of 2 entries

Previous

1

Next

The dialog displays the following fields: Extension (number) Boss/Secretary (status) Secretary Extension (number) Personal Assistant (status) Follow-me (status) DND (Do Not Disturb status) Call Forward Immediately (CFI status) Call Forward Busy (CFB status) Call Forward No Answer (CFN status) Call Forward Unavailable (CFU status) Call Completion (status).

Use the (✎) button that is located at the end of each line to modify the status of an extension. You can modify the Personal Assistant and Diversion settings of the extension. The status can be in effect either unconditionally (by NOT selecting a time group from the time dropdown), or for a specific during the time period as defined by the time group that you select from the dropdown list, or can be effect unconditionally if no time group is selected. In addition, you can change the status by click the circle in each option.

Extension Status

Personal Assistant

The Personal Assistant IVR will not be activated until a greeting has been recorded.

Press 1	Select Module	Select Destination
Press 2	Select Module	Select Destination
Press 3	Select Module	Select Destination
Press 4	Select Module	Select Destination
Default	Select Module	Select Destination

Boss/Secretary	-- Select Time --	No
Follow-me	-- Select Time --	No
DND	-- Select Time --	No
Call Completion	-- Select Time --	No

CFI	Select Module	Select Destination	-- Select	No
CFB	Select Module	Select Destination	-- Select	No
CFN	Select Module	Select Destination	-- Select	No
CFU	Select Module	Select Destination	-- Select	No

Close

Save

Use the (📞) button see the status of the devices associate at the extension.

Devices Info

Username / Channel	Host	Port	Status	User Agent	Description
PJSIP/1000	(Unspecified)	0	---		TESTE

Close

5.2 Applications

5.2.1 Conferences

A conference room allows a group of people to participate in phone call. The most common form of bridge allows participants dial into a virtual meeting room from their own phone. Meeting rooms can hold dozens or even hundreds of participants. This contrasts with three-way calling, a standard feature of most phone systems which only allows a total of three participants. For most phone systems, conference bridging is an add-on feature those costs thousands of dollars.

General

Extension, number to dial to reach this service. This is a number that internal endpoints can dial to reach this conference. Like the ring groups, this can be thought of as the extension number of the conference.

Description, short description for identify this conference.

Max Members, this option limits the number of participants for a single conference to a specific number. After the limit is reached, the conference will be locked until someone leaves. Note however that an Admin user will always be allowed to join the conference regardless of if this limit is reached or not.

Video Mode, Options:

- None: No video sources are set by default in the conference. It is still possible for a user to be set as a video source via AMI or DTMF action at any time.
- Follow Talker: The video feed will follow whoever is talking and providing video.
- Admin: The first administrator who joins the conference with video capability is the only source of video distribution to all participants. If the administrator leaves, the next administrator to join after them becomes the source.

User PIN*, this is a numeric passcode that is used to enter the conference room. If a PIN is entered in this field, no one can join the conference room without entering the PIN.

Admin PIN, functions in the same way as the User PIN. The Admin PIN and User PIN should not be set to the same value. The Admin PIN is used in conjunction

with the Wait Admin option explained further in this chapter, to identify the administrator or leader of the conference.

Language, here you can set the language used for voice prompts to the conference. **Record Conference**, when set to yes, records the conference call starting when the first user enters the room, and ending when the last user exits the room.

Conference Settings section

Music on Hold, the music on hold class to use for this conference.

Announce User Count, used for announcing the participant count to all members of the conference. If set to a number, then the announcement is only played when the number of participants is above the set number. Available options are yes, no, or a whole number. Default is no.

Music on Hold When Empty, when this option is enabled on-hold music will be played if there is only one caller in the conference room or if the conference has not started yet (because the leader has not arrived). If this option is disabled, no sound will be played during these situations.

User Count, when this option is enabled, the number of users currently in the conference room will be announced to each caller before they are bridged into the conference.

Announce Join/Leave, when enabled, this option will prompt the user for a name when entering the conference. After the name is recorded, it will be played when the user enters or exits the conference.

Announce Only User, sets if the only user announcement should be played when a user enters an empty conference.

Wait for Leader, if this option is enabled, the conference will not begin until the conference administrator joins the conference room. The administrator is identified by the Admin PIN. If other callers join the conference room before the leader does, they will hear on-hold music or silence until the conference begins (what they hear depends on the MoH When Empty setting explained earlier in this section). If this option is set to "No", the callers will be bridged into the conference as soon as they call the conference room number.

Start Muted, when this option is enabled, all users joining the conference are initially muted.

Drop Silence, this option drops what Communications Systems detects as silence from entering the bridge. Enabling this option will drastically improve performance and help remove the buildup of background noise from the conference. Highly recommended for large conferences due to its performance enhancements.

Quiet, when this option is enabled, user introductions, enter prompts, and exit prompts are not played. There are some prompts, such as the prompt to enter a PIN number that will still be played regardless of how this option is set.

Kick Users, enabling this option will kick out all remaining users of the conference, after the last admin user leaves the conference.

Talk Detection, this option sets whether notifications of when a user begins and ends talking should be sent out as events over AMI.

Allow to Invite, if enabled, all the participants could press "***" or "0" to invite other people to this conference.

The following codes can be entered by all conference participants:

***1** – toggles mute for the user. When enabled, anything the user says is not transmitted to the rest of conference members. If the conference is being

recorded, anything said by a muted user is not part of the recording.

***4** - decreases receive volume. The user can tap this option to decrease the volume of what they are hearing. This does not affect what any other conference member hears. If a user is finding other conference members too loud, they can press *4 a few times to make the conference quieter for themselves.

***5** - increases receive volume. The user can tap this option to increase the volume of what they are hearing. This does not affect what any other conference member hears. If a user is having trouble hearing other members of the conference, they can press *5 a few times to make the conference louder for themselves.

***6** - decreases transmit volume. The user can tap this option to decrease the volume of what they are transmitting to the rest of the conference members. When this option is used, the user will sound quieter to all other conference members. If a user is much louder than the other members of a conference room, they can tap *6 a few times to make them transmit volume quieter.

***7** - increases transmit volume. The user can tap this option to increase the volume of what they are transmitting to the rest of the conference members. When this option is used, the user will sound louder to all other conference members. If the conference members are having trouble hearing a particular user, that user can tap *7 a few times to make them transmit volume louder.

***8** – user can tap this code to leave the conference.

In addition, the admin has access to additional codes:

***2** - toggles the conference lock. When a conference is locked, no more callers may join. A locked conference must be unlocked for any new users to join. This option is only available to a conference administrator. If the conference does not have an admin PIN configured or the user has joined the conference as a user instead of an admin, this option is not available.

***3** - Kicks the last user who joined the conference from the conference room. The user will hear a message informing them that they have been kicked from the conference and that their call will be terminated. Note that if a conference is unlocked, the user may rejoin. The best way to remove an abusive conference user is to eject them and then immediately lock the conference. This option is only available to a conference administrator. If the conference does not have an admin PIN configured, or the user has joined the conference as a user.

5.2.2 Custom Applications

This module allows you to call modules that have no extension number, such as pre-announcement, Time Condition, IVR, etc. A custom application is a custom feature code. A custom application allows a custom extension or star code to be defined, which will direct the caller to any call target when dialed. For example, if we have a ring group that calls the cell phones of all staff members, we might create a custom application that calls that ring group when CELL (2355) is dialed.

General

The screenshot shows a web interface for configuring a custom application. At the top, there's a tab labeled 'GENERAL' and a menu icon. The form contains the following elements:

- Code ***: A text input field.
- Name ***: A text input field.
- Enabled**: A toggle switch currently set to 'Yes'.
- Destination ***: A section containing two dropdown menus: 'Select Module' and 'Select Destination'.
- Save**: A green button at the bottom right.

Code *, number to dial to reach this service.

Name*, short description to identify this custom application.

Enabled, enable or disable this custom application. If disabled, then users will be informed that the extension they dialed is not valid if they attempt to use the custom application. This field allows a custom application to be quickly disabled without having to remove the application entirely.

Destination* section

Select Module, allows to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed. Any call target that has been previously configured is a valid destination for a custom application.

5.2.3 Custom Destinations

A custom destination is used to add a custom call target that can be used by Attimo CloudPBX dialogs. Anything that can be dialed from a user's extension can be turned into a custom destination. For example, by default, there is no way to send an inbound caller directly to the messaging center so that the caller could log in and check their voicemail messages. For example, A custom destination could be set up to dial *98 and then an inbound route could point directly to that custom destination. A caller who was routed through that inbound route would immediately hear the prompts to log into their voicemail box, just as if they were a user on the PBX and had dialed *98.

General

Description*, is used to identify this destination when it is being selected as a call target in other dialogs.

Number to Dial*, is the extension, telephone number, or feature code that the system should dial when a caller is routed to this destination. Anything that can be dialed from a user's extension can be entered into this field.

Class of Service*, Class of Service to search the number to dial.

GENERAL

Description *

Caller ID

Name

Number

Number to Dial *

Class of Service *

All Permissions

Save

5.2.4 Custom Context

Allows the integration of personalized contexts, especially useful for advanced users.

GENERAL

Code *

Name *

Enabled

Yes

Destination *

Select Module

Select Destination

Save

Description*, a short description to identify this custom context.
Context*, name of the custom context created by yourself. **Extension**, the extension defined in your custom context. **Priority**, the priority defined in your custom context.
Destination, destination after having executed the custom context.

5.2.5 Feature Codes

Attimo CloudPBX includes all the telephony features currently available in all Communications Systems distributions plus features than until now were only available in expensive, commercial PBX systems.

Blacklist Section

Prompts the user to enter a telephone number. The entered number is then added to the user's blacklist. Inbound calls will not ring an extension if they are on that extension's blacklist. Blacklisted callers will be told that the number they dialed is no longer in service. This option is very popular as it allows users to block unwanted numbers.

Blacklist

Blacklist a Number	*30	DefaultCustom	EnabledDisabled
Remove Number From Blacklist	*31	DefaultCustom	EnabledDisabled
Blacklist Last Caller	*32	DefaultCustom	EnabledDisabled

***30 - Add Number to Blacklist**, prompts the user to enter a telephone number. The entered number is then added to the user's blacklist. Inbound calls will not ring an extension if they are on that extension's blacklist. Blacklisted callers will be told that the number they dialed is no longer in service.
***31 - Remove Number from Blacklist**, prompts the user to enter a telephone number. The entered number is removed from the user's blacklist.
***32 - Add Last Caller to Blacklist**, adds the last number that called the user to the blacklist.

Business Services section

Here we can find the following feature codes:

Business Services

Wakeup Call	*34	DefaultCustom	EnabledDisabled
Remote Wakeup Call	*35	DefaultCustom	EnabledDisabled
Speak Last Number	*37	DefaultCustom	EnabledDisabled
Reminder	*38	DefaultCustom	EnabledDisabled

***34 - Wakeup Call**, set up a reminder or wakeup call for the current extension.

***35 - Remote Wakeup Call**, create a reminder or wakeup call for another extension.

***36 - Boss/Secretary**, this functionality is used to re-route all incoming calls for the boss phone to the secretary phone. Only the secretary is allowed to call the boss phone directly.

***37 - Speak Last Number**, says the last number that called the current extension, with the possibility to press a button to call back to the original caller.

***38 - Reminder**, records a message. You can configure in how many minutes you want to hear the recording. When the set time expires, you will receive a call to your extension and the recording will be played.

Call Completion Section

Call Completion Supplementary Services (often abbreviated "CCSS" or simply "CC") allows a caller to let Attimo CloudPBX automatically alert him when a called party has become available, given that a previous call to that party failed for some reason. The two services offered are Call Completion on Busy Subscriber (CCBS) and Call Completion on No Response (CCNR). To illustrate, let's say that Alice attempts to call Bob. Bob is currently on a phone call with Carol, though, so Alice hears a busy signal. In this situation, assuming that Communications Systems has been configured to allow for such activity, Alice would be able to request CCBS. Once Bob has finished his phone call, Alice will be alerted. Alice can then attempt to call Bob again.

Call Completion (CCSS)

Call Completion - Toggle	*40	DefaultCustom	EnabledDisabled
Cancel Call Completion	*41	DefaultCustom	EnabledDisabled

***40 - Call Completion**, to activate a call to a previously unresponsive extension when that extension become reachable.

***41 - Cancel Call Completion**

Call Center section

Call centers are special offices that are purpose-built to handle a large volume of phone calls. Call centers typically handle customer service, support, telemarketing, billing and collection functions. The employees who staff call centers are referred to as "agents" or "customer service representatives". Call centers range from very small informal operations to quite large, highly optimized sites with hundreds of agents. This group of feature codes allows us to interact with various features directly from the phone:

Call Center

Add/Remove Queue Agent	*50	Default	Custom	Enabled	Disabled
Pause/Unpause Queue Agent	*51	Default	Custom	Enabled	Disabled
Queues Login/Logout	*52	Default	Custom	Enabled	Disabled
Queues Pause/Unpause	*53	Default	Custom	Enabled	Disabled
Spy on Extension In Barge Mode	*54	Default	Custom	Enabled	Disabled
Spy on Extension	*55	Default	Custom	Enabled	Disabled
Spy on Extension In Whisper Mode	*56	Default	Custom	Enabled	Disabled
Spy Random Channels	*57	Default	Custom	Enabled	Disabled
Hot Desking	*80	Default	Custom	Enabled	Disabled
Hot Desking CC	*90	Default	Custom	Enabled	Disabled

***50 - Add/Remove Queue Agent**, Add/Remove an Agent to a specific queue.

***51 - Pause/Unpause Queue Agent**, Pause/Unpause an Agent to a specific queue.

***52 - Queues Login/Logout**, Add/Remove an Agent to all queue that agent belong to.

***53 - Queues Pause/Unpause**, Pause/Unpause an Agent to all queue that agent belong to.

***54 - Spy on Extension in Barge Mode**, this will create a three-way conference between the client, the agent, and the user barging in.

***55 - Spy on Extension**, spy on a specified extension.

***56 - Spy on Extension in Whisper Mode (Coaching)**, spy on a specified extension in whisper mode.

***57 - Spy Random Channels**, spy on random channels.

Call Forward section

Call Forward (CF)

Boss/Secretary - Toggle	*36	Default	Custom	Enabled	Disabled
Call Forward Immediately - Toggle	*58	Default	Custom	Enabled	Disabled
Set CF Immediately Number	*59	Default	Custom	Enabled	Disabled
Call Forward Unavailable - Toggle	*60	Default	Custom	Enabled	Disabled
Set CF Unavailable Number	*61	Default	Custom	Enabled	Disabled
Call Forward Busy - Toggle	*62	Default	Custom	Enabled	Disabled
Set CF Busy Number	*63	Default	Custom	Enabled	Disabled
Call Forward On No Answer - Toggle	*64	Default	Custom	Enabled	Disabled
Set CF On No Answer Number	*65	Default	Custom	Enabled	Disabled
Do Not Disturb - Toggle	*66	Default	Custom	Enabled	Disabled
Follow Me - Toggle	*67	Default	Custom	Enabled	Disabled
Clear all Diversions	*69	Default	Custom	Enabled	Disabled
Personal Assistant - Toggle	*96	Default	Custom	Enabled	Disabled

The group of call forwarding provides the following options:

- *36 - Boss/Secretary – Toggle**, enable or disable call forwarding from the boss to the secretary.
- *58 - Call Forward Immediately**, enable or disable call forwarding.
- *59 - Set CF Immediately Number**, set the number to which diverted calls should be sent.
- *60 - Call Forward Unavailable**, enable or disable call forwarding.
- *61 - Set CF Unavailable Number**, set the number to which diverted calls should be sent.
- *62 - Call Forward Busy**, enable or disable call forwarding when your extension is busy.
- *63 - Set CF Busy Number**, set the number to which diverted calls should be sent.
- *64 - Call Forward on No Answer**, enable or disable call forwarding when your extension does not answer incoming calls.
- *65 - Set CF On No Answer Number**, set the number to which diverted calls should be sent.
- *66 - Do Not Disturb**, enable or disable Do Not Disturb.
- *67 - Follow Me**, enable or disable Follow Me.
- *69 - Clear all Diversions**, disable all diversions.
- *96 - Personal Assistant – Toggle**, enable or disables the Personal Assistant for the extension.

On Call Features section

On Call Features

Disconnect Call	*0		
Pause/Resume Outbound Call Recordings	*00		
Direct Pickup	*07	<input checked="" type="button" value="Default"/> Custom	<input checked="" type="button" value="Enabled"/> Disabled
Pickup Group	*08	<input checked="" type="button" value="Default"/> Custom	<input checked="" type="button" value="Enabled"/> Disabled
Pause/Resume Incoming Call Recordings	*1		
Attended Transfer	*2		
One Touch Recording	*3		
Park Call	*4		
Blind Transfer	#1		

These facilities are used when you are on a call:

- *0 - Disconnect Call**, disconnect the current call.
- *07 - Direct Pickup**, remotely capture a call that is ringing at another extension.
- *08 - Pickup Group**, captures any call that is ringing the group that belongs to the extension if it has the right permission. To use this facility is necessary to create a pickup.
- *2 - Attended Transfer**, transfer the current call with notifying the extension to which the call will be transferred.
- *3 - One Touch Recording**, forces the call to be recorded.
- *4 - Park Call**, call parking.
- #1 - Blind Transfer**, transfer the current call without notifying the extension to which the call will be transferred.

Phonebook Directory Section

This directory is completely linked to extensions that have voice mail, with which they create a name, and the full name of the person is recorded to listen when there is a match when the user will type the first few letters of the name or last name.

Phonebook Directory

Dial By Name Directory	411	Default	Custom	Enabled	Disabled

411 - Dial by Name Directory

Test Services section

Test Services

Speak Date and Time	*70	Default	Custom	Enabled	Disabled
Speak Your Extension Number	*71	Default	Custom	Enabled	Disabled
Echo Test	*72	Default	Custom	Enabled	Disabled
Simulate Incoming Call	*73	Default	Custom	Enabled	Disabled

These are a group of features to test the system.

***70 - Speak Date and Time, Listen to the current date and time.**

***71 - Speak Your Extension Number**, you can hear your extension number.

***72 - Echo Test**, an echo test system to measure the response time.

***73 - Simulate Incoming Call**, simulation of an incoming call to test ringing of the phone.

Special Features Section

Special Features

Lock/Unlock Phone	*75	Default	Custom	Enabled	Disabled
Change Features Password	*76	Default	Custom	Enabled	Disabled
Remote Substitution	*77	Default	Custom	Enabled	Disabled
Customer Code	*78	Default	Custom	Enabled	Disabled
Authorization Code	*79	Default	Custom	Enabled	Disabled
Night Mode All	*81	Default	Custom	Enabled	Disabled
Paging	*82	Default	Custom	Enabled	Disabled
Paging Duplex	*83	Default	Custom	Enabled	Disabled
Anonymous Calling	*88	Default	Custom	Enabled	Disabled

This is a group of features which are described below:

***75 - Lock/Unlock Phone**, lock and unlock the current extension.

***76 - Change Features Password**, change the password to access to certain telephone facilities.

***77 - Remote Substitution**, makes it possible for a remote phone to make calls as if you are on your own phone.

***78 - Account Code**, CDR assigned to an account code, very useful for call accounting.

***79 - Authorization Code**, from any phone you can make a call using a code that is associated with unrestricted dial plan.

- *80 - Hot Desking**, assign an extension number to a hot desking device type.
- *81 - Night Mode All**, change the state of all-night mode.
- *82 - Paging**, places an outgoing call from the given extension, and returns the call on conference as a muted participant, e.g. 827000. The feature must be enabled in the Feature Category associated with the extension.

Recording & Announcements Section

This group of feature codes allows you to interact with your voice mail and other similar applications.

Recordings & Announcements

Custom Recording	*92	Default	Custom	Enabled	Disabled
Dictation	*93	Default	Custom	Enabled	Disabled
Record Msg For Personal Assistant	*94	Default	Custom	Enabled	Disabled
Send Voicemail Message	*95	Default	Custom	Enabled	Disabled
Direct Voicemail	*97	Default	Custom	Enabled	Disabled
Remote Voicemail	*98	Default	Custom	Enabled	Disabled

- *92 - Custom Recording**, records a message.
- *93 - Dictation Services**, records a message with the option of sending it by email.
- *94 - Record Msg for Personal Assistant**, records a message that callers will hear when they are served by the personal assistant.
- *95 - Send Voicemail Message**, allows you to dial any extension and leave a voicemail. For example, when dialing *95*2492, you can leave a voicemail message to extension 2492. The feature must be enabled on the Feature Category associated to the extension.
- *97 - Direct Voicemail**, direct entry to the voicemail system – requires password.
- *98 - Remote Voicemail**, remote entry to the voicemail – requires both extension number and password.

5.2.6 Paging & Intercom

This module creates hunt groups to which they were able to send a message marked a number. It can also be used to create an intercom between 2 extensions.

General

Code*, number to reach this service.

Description*, short description to identify this paging group.

Extensions*, list of the extension(s) to dial.

Announcement*, announcement to be played to all paged participants.

Timeout, this specifies the length of time that the system will attempt to connect a call. After this duration, any intercom calls that have not been answered will be hung up by the system.

Mode, it allows you to define the paging behavior, Options:

- Default: Default behavior. Plays the announcement if defined and then allows to the caller continue speaking.
- Announcement Only: Plays the announcement and then hang

up. **Duplex Audio**, sometimes referred to as "talkback paging." The use of this option implies that the equipment that receives the page has the ability to transmit audio back at the same time as it is receiving audio. Generally, you would not want to use this unless you had a specific need for it.

Ignore Forward Call, ignore attempts to forward the call.

Quiet Mode, it does not play a beep to caller.

Record Paging, this allows you to record the page message into a file.

Skip Busy, dials a channel only if the device state is NOT_INUSE. This option is likely only useful (and reliable) on SIP-bound channels, and even so may not work if a single line is allowed multiple calls on it.

5.2.7 Pickup Groups

Pickup Group specifies to which pickup groups an extension belongs. An extension can belong to multiple pickup groups. The call group and pickup group options allow users to pick up calls that are not directed to them by dialing a feature code (*08). Calls directed to any phone in a particular call group can be answered by any user who is a member of the corresponding pickup group. For example, a user in pickup group Support will be able to pick up any call directed to any phone in the Support call group. This can be useful for small office or home setups, where it is easier to simply pick up a call from another phone rather than forward that call to another extension. Note that a user can be part of a pickup group without being a member of the associated call group. For example, a senior staff member may be able to pick up any call directed to anyone in his department, but his department should not be able to pick up calls directed to the senior staff member.

General

GENERAL

Description *

Members *

Extension	Member	Allow Pickup
Add		

Description*, a description to identify the pickup group, can consist of numbers and names.

Extension Section

Extension, any extension that you want to add to this group.

Member, when enabled, indicates that the extension is a member of the pickup group.

Allow Pickup, when enabled, indicates that this extension can pick up calls that are directed to this group.

5.2.8 Parking

The Call Parking feature allows you to place a caller on hold and then retrieve them from any phone, anywhere on the system.

General

GENERAL

Code *		Call Transfer	No
Description *		Call Reparking	No
Parking Positions	10	Call Hangup	No
Parking Time	45	Find Slot	Next
Comeback Dial Time	30	Return to Originator	Yes
Courtesy Tone	Caller	Announce Space Number	Yes
Music on Hold	Default	Record	No

Timeout Destination *

Save

Code*, number to reach this service.

Description*, short Description to identify this parking.

Parking Positions, number of parking spaces to use.

Parking TimeOut*, number of seconds a call can be parked before being returned. **Comeback DialTime***, when a parked call times out, this is the number of seconds to dial the device that originally parked the call.

Courtesy Tone, to whom to play the courtesy tone while waiting for someone to pick up the parked call. Options are:

- No one
- Caller
- Callee
- Both

Music on Hold, this is the MoH class to use for the parked channel.

Call Transfer, enables or disables DTMF based transfers when picking up a parked call.

Call Re-Parking, enables or disables DTMF based parking when picking up a parked call.

Call Hang up, enables or disables DTMF based hang up when picking up a parked call.

Find Slot, sets the method for selecting parking spaces when a call is parked. Options are:

- First: use the lowest numbered parking space available
- Next: use the next parking space from the most recently used

one. **Return to Origin**, setting this option configures the behavior of call parking when the parked call times out

Announce Space Number, if set to no, the announcement of the parking space number will be silenced.

Timeout Destination section

Select Module, select the module to used when a parked call timed out.

Select Destination, destination for a parked called that has timed out.

5.2.9 Speed Dialing

This feature which permits fast dialing of frequently used numbers. Sometimes there are long numbers we want to abbreviate a short dialing.

General

GENERAL

Description *		Destination Number *	
Speed Dial Code *		Class of Service	Inherited

Save

Speed Dial Code*, number to access the speed dial. This number must be unique.

Description*, short Description to identify this Speed Dial.

Class of Service, class of service to use for dialling this speed dial.

Dial Number*, the number that will be dialed by this speed dial.

5.2.10 Import/Export Speed Dialing

Import/Export Speed Dialing to/from CSV file.

General

GENERAL

CSV File

Import Speed Dials

Export Speed Dials

An example of the file format can be download by press the “Download Import Format” button at the bottom of the screen. The first row is for the header.

	A	B	C	D	E
1	mode	speeddial	dest_number	description	class_of_service
2	add	*6969	8264	test	
3					

5.2.11 Voicemail Broadcast Group

Extension Group to which voicemail can be sent.

General

GENERAL

Code*

Password

Description*

Skip Instructions

Extensions*

No

Save

- Code***, number to dial to send broadcast voicemail.
- Description***, short Description to identify this Voicemail Broadcast Group.
- Extensions***, list of extensions for mass-mailing voicemail. Note that you can only see the extensions that have active voicemail.
- Password***, password to protect this Voicemail Broadcast Group.
- Skip Instructions**, if enabled, skips the playback of instructions for leaving a message to the listed extensions on this VM group.

5.2.12 Call Back

Callback is a call target that will immediately hang up on a caller, call them back, and then redirect the call to another call target. This is most often used to avoid long- distance charges for remote agents who do not have access to a VoIP endpoint. This is especially relevant in the case of mobile phones where incoming calls are usually significantly cheaper than outgoing calls. The callback target may connect the caller with any resource on Attimo CloudPBX (such as an extension, the voicemail messaging center, or a queue), or it may be used in conjunction with DISA to give the caller a dial tone on the system from which they can call any telephone number they wish.

General

The screenshot shows the 'GENERAL' configuration page for a Call Back target. The page has a header with 'GENERAL' and a menu icon. The main content area contains several fields: 'Description *' (text input), 'Number' (text input), 'Dial Prefix' (text input), 'Delay' (text input with value '5'), 'Class of Service' (dropdown menu with 'All Permissions' selected), and 'Destination *' (two dropdown menus: 'Select Module' and 'Select Destination'). A green 'Save' button is located at the bottom right of the form.

Description*, short Description to identify this callback.

Number, this is the phone number that Attimo CloudPBX will dial to reconnect with the caller after the call that initiated the callback is terminated. The number must be in a format that one of the outbound routes configured in the Outbound Routes section of Attimo CloudPBX can be matched with (for example, if there is no outbound route defined to match a 10-digit dialing pattern, entering 5551234567 for this field would render the callback configuration useless, as the outbound callback would never be completed). If the field is left blank, then Attimo CloudPBX will attempt to call back the caller ID number that initiated the callback.

Dial Prefix, this prepends the prefix to the number to dial.

Delay, delay before return call. **Class of Service**, Class of Service for making the call.

Destination* Section

Select Module, to choose which module should be activated.

Select Destination, to configure the call target that the caller will be connected to once the callback dialog reconnects the caller to Attimo CloudPBX. Any existing call target can be used.

A few common examples of when a callback target might be used are as follows:

A company where employees need the ability to check their voicemail from anywhere. Calling the toll-free company phone number costs the company too much money. A callback target could be set up to call back the incoming caller ID and be directed to the miscellaneous destination of *98. Callers would receive a call on the number they called from, would be prompted for their extension and

their password, and would then have access to their voicemail messages.

A company receives better per-minute rates on calls made through its VoIP trunks than calls made through employees' mobile phones. Employees' mobile phones have free incoming calls. A callback target could be set up for each employee with a mobile phone to call back the employee's mobile number. The callback would be directed to a DISA destination to give the employee a dial tone on the PBX (allowing them to dial out using the company's VoIP trunks without using any outgoing mobile minutes).

A company that receives collect calls from anywhere in the world (such as a credit card company that needs to receive calls if a customer's card is lost or stolen). The company reduces their costs if they use a VoIP trunk local to the country that the customer is in, rather than paying for the entire collect call at hefty international rates. A callback target could be set up to call back the incoming caller ID of the customer and be directed to a queue. The customer would receive a call to the number they called from and would relate to a company representative as soon as one is available.

5.2.13 DISA

DISA allows you to create a destination that allows people to call in to from an outside line and reach the system dial tone. This is useful if you want people to be able to take advantage of the low rate for international calls that you have available on your system, or to allow outside callers to be able to use the paging or intercom features of the system. Always protect this feature with a strong password.

A DISA call target will provide a caller with a dial tone on Attimo CloudPBX. Once the caller has a dial tone, they can utilize the same set of functions that are utilized by a user with VoIP endpoint attached to Attimo CloudPBX. This means that a person who is remotely located could be given access to dial any extension directly, check their voicemail messages, or even place calls to external telephone numbers through Attimo CloudPBX.

General

The screenshot shows the 'GENERAL' settings tab for a DISA target. The form includes the following fields:

- Description ***: A text input field.
- Password**: A text input field containing the value '64484'.
- Class of Service**: A dropdown menu with 'All Permissions' selected.
- Caller ID**: A section with two sub-fields: 'Name' and 'Number'.
- Response Timeout**: A text input field containing the value '10'.
- Digit Timeout**: A text input field containing the value '5'.

A green 'Save' button is located at the bottom right of the form.

Description*, short description for identify this DISA when it is being selected as a call target in other parts of the Attimo CloudPBX interface.

Password, this is used to authenticate a caller when they want to activate DISA feature. If the password field contains a value, then the caller will be prompted to enter the password. The password that the user enters must match the value of the password field, otherwise the call will be disconnected, and the caller will not be able to access the DISA feature.

Class of Service, this specifies the dial plan context in which the user-

entered extension will be matched.

Caller ID, this is used to set the outbound caller ID, consisting of two parts: the CID Name and the CID Number. This will define the caller ID text that is displayed when this user calls other. This is an optional field. If this field is left blank, then the caller ID of the person placing the call will be used.

Response Timeout, this specifies how long Attimo CloudPBX will wait for valid input before disconnecting the call. This not only applies when a caller has not entered any digit yet, but also if a caller has partially entered a number to call without finishing the entry. The default value for this field is 10 seconds.

Digit Timeout, this specifies how long Attimo CloudPBX will wait between digits before dialing the call. If a caller begins entering digits and then stops, Attimo CloudPBX will wait for the number of seconds specified in this field, before sending the entered digits to Attimo CloudPBX for dialing. The default value for this field is five seconds. This is usually sufficient as most people do not take more than five seconds between button pushes on their phone once they have started dialing.

5.2.14 PIN List

List of PIN that is associated with an outgoing trunk. Any extension that wants to make a call you will be asked for a PIN number.

General

GENERAL

Description *

PIN List

Save

Description*, this is a short description for this PIN List.

Pin List*, here you set the PIN List for this record (If you want to add more than one pin do it in a new line). You can only introduce numbers and the characters * and #.

5.2.15 Dynamic Destinations

With this add-on, you can dynamically route the calls based on the DID. You can perform a query to a database or API, and depending on the response, route the number to a specific destination.

GENERAL

Description *

Source Type

MySQL

Database *

Query *

Default Destination *

Select M... Select D...

Host *

localhost

Username *

root

Password

Destinations

Response	Destination	Enabled
	Select Module	Select Destination
		Yes Add

Save

Here are the different options that can be configured.

For MySQL Source Type

Description, a brief description for this Dynamic Destination.

Database, MySQL database to query.

Query, Query to perform. The special token [CIDNUM] will be replaced by the CID number of the calling party, for example, *SELECT 'VIP_CUSTOMER'; as 'response' FROM 'customers' WHERE 'number' = '[CIDNUM]'*

Default Destination, destination to use in case we receive an un-defined answer, or not taken into consideration for the destinations configured.

Host, IP address or hostname for the server where the MySQL database is located at.

Username, database username used to perform the query.

Password, database user password used to perform the query.

For URL Source Type

Description, a brief description for this Dynamic Destination.

Host, IP address or hostname for the server where the script is located at.

Path, name of the script to execute.

Query String, Series of parameters that will be sent as part of the script. The special argument [CIDNUM] will be replaced with the CID number of the calling party. For example, *"caller_num=[CIDNUM]&ctype=vip"*

Default Destination, destination to use in case we receive an un-defined answer, or not taken into consideration for the destinations configured.

Port, port used to perform the request. Normally 80 for HTTP, and 443 for HTTPS.

Auth User, user to authenticate the HTTP/HTTPS request.

Auth Password, Password used to authenticate the HTTP/HTTPS request.

Secure, if enabled, the request will be sent through HTTPS, else it will be sent through HTTP.

Destinations

Response, the expected response from the query. **Destination,** the destination to which we route the call to. **Enabled,** enable or disable the routing based on the response.

5.3 Class of Service

5.3.1 Class of Service

Class of Service is a Group of settings that define the dial plan that has access to the extension.

General

GENERAL

Class of Service *

Description *

Feature Category **All Features** ▼

Dial Restrictions **No Dial Restrictions** ▼

Route Selection **-- All Outbound Routes --** ▼

Allowed Calls By

Private ☐ No

Save

Class of Service *, Class of Service Name (Must be Unique). Alphanumeric values with dash and underscore are allowed.

Description *, this is a short Description for identify this Class of Service.

Feature Category, features allowed for this Class of Service.

Dial Restrictions, dial restriction rules set for this Class of Service.

Route Selection, routes to use for this Class of Service.

Allowed Calls By, it defines the list of CoS to be allowed to call to this CoS when Private field is checked.

Private, it defines if extensions with this CoS may be called by others with different CoS. If is checked only calls with the same CoS or calls coming from CoS selected on Allowed Calls By field will be allowed. It applies for internal calls only.

5.3.2 Feature Categories

In this module you can create groups of feature codes. This allows you to prevent some users from having access to some of the more sensitive feature codes.

General

GENERAL

Description *

Available Features

Add all

#1 - Blind Transfer	+
*0 - Disconnect Call	+
*00 - Pause/Resume Outbound Call Recordings	+
*07 - Direct Pickup	+
*08 - Pickup Group	+
*1 - Pause/Resume Incoming Call Recordings	+
*2 - Attended Transfer	+
*3 - One Touch Recording	+
*4 - Park Call	+
*30 - Blacklist a Number	+
*31 - Remove Number From Blacklist	+
*32 - Blacklist Last Caller	+
*34 - Wakeup Call	+
*35 - Remote Wakeup Call	+

Enabled Features

Remove all

Save

Description*, this is a short description to identify this feature category - must be unique.

Available Features, list of available feature codes.

Enabled Features, list of feature codes that have been included.

5.3.3 Dialing Restriction Rules

Here you can create dial restrictions rules. These can be associated with a class of services.

General

GENERAL

Description *

Custom Rules Context

Outbound Restriction Rules

Rule (Pattern)	Allowed	Announcement	Play Max Duration	Max Duration	Password
Pattern to Match	No	None	No	Max duration in seconds	No
Add					

Internal Restriction Rules

Rule (Pattern)	Announcement
Pattern to Match	None
Add	

Save

Description*, short description to identify this "Dialing Restriction" - must be unique.
Custom Rules Context, this allows you to include a custom context with your own advanced rules.

Rules section

Rules (Pattern), allows you to create extension patterns in your dial plan that match one or more possible dialed numbers. The pattern options are:

- The letter X or x represents a single digit from 0 to 9.
- The letter Z or z represents a single digit from 1 to 9.
- The letter N or n represents a single digit from 2 to 9.
- The period (.) character is a wildcard that matches one or more characters.
- The exclamation mark (!) is a wildcard that matches zero or more characters.
- [1237-9] matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9)
- [a-z] matches any lower-case letter.
- [A-Z] matches any UPPER-case letter.

Allowed, allow/disallow this pattern.

Announcement, you can choose an announcement associated with this pattern.

Play Max Duration, play max duration only if max duration is greater than 0 or not

in blank.

Max Duration, maximum call duration associated with this pattern.

Password, this determines if a password is required when using this pattern.

5.3.4 Customer Codes

Customer codes associated dynamically to a call-in order to register this code in the CDR.

General

GENERAL

Customer Code *

Description *

Enabled

Yes

Save

Customer Code*, customer code number.
Description*, short description to identify this Customer code.
Enabled, enable/disable this Customer Code.

5.3.5 Authorization Codes

Code that grants privileges to make a call from any extension, based on its Class of Service.

General

GENERAL

Code *

Alias *

Description *

Class of Service *

All Permissions

Enabled

Yes

Save

Authorization Code*, authorization code number.
Authorization Alias*, Alias to identify this authorization code. For security reasons this will appear on CDR reports instead of the authorization code.
Description*, Short description to identify this "Authorization Code".
Class of Service, Class of Service is used to route the call.
Enabled, enable/disable this Authorization Code.

5.3.6 Route Selections

Route selection is a private branch exchange (PBX) feature that allows a system to route a telephone call over the most appropriate carrier and service offering based on factors such as the type of call (i.e., local, local long distance, etc.), the user's class of service (CoS), the time of day, and the day of the week (e.g., workday, weekend, or holiday).

General

GENERAL

Description *

Route Selection Members

	Outbound Route	Time Group	Enabled	
	-- Select Outbound Route --	-- Select Time --	Yes	
Add				

Save

Description*, a short description for identify this Route Selection.

Route Selection Members section

Outbound Route, here you can select outbound route.

Time Group, here you can select time group.

Enabled, enable or disable this route.

5.4 Call Center

5.4.1 Ring Groups

Ring Groups offer the possibility that a call to be received by more than one internal extension. The option is used most often for picking up calls received on a certain line (Inbound Routes) and sending them to a certain destination (Welcome prompt, IVR and ring groups). Also, the group can be accessed internally, calling the code assigned to it.

General

GENERAL

Code *

Description *

Extensions

External Numbers

Ring Strategy

Ringall

Ring Time

Default (30)

Class of Service

All Permissions

Ringback Tone

None (Ringback)

CID Name Prefix

Announcement

None

Allow Diversions

No

Mark Cancelled Calls as Answered

No

Answer Channel

Yes

Skip Busy Extensions

No

No Release

Yes

Last Destination *

Select Module

Select Destination

Save

Code*, number to dial to reach this service.

Description*, short description to identify this ring group.

Extensions, list of extension for this ring group.

External Numbers, list of external number for this ring group.

Ring Strategy, ring strategy of extension group.

Ring Time, ring time in seconds, MAX 160 seconds.

Class of Service, Class of Service to use for dialing the external numbers listed in this ring group.

Music on Hold, music on hold to play.

CID Name Prefix, prefix to append to this ring group.

Allow Diversions, allows you to define if the diversions defined for the different extension members will be applied or not.

Mark Cancelled Calls as Answered, with this option enabled prevents the other phones to record a missed call when the call has been answered by another member listed on this ring group. This is a very useful setting when the ring

strategy is set to "Ring All".

Last Destination Section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

5.4.2 Queues

ACD (Automatic Call Distributor) is what distributes incoming calls in the order of arrival to the first available agent. The system answers each call immediately and, if necessary, holds it in a queue until it can be directed to the next available call center agent. Balancing the workload among agents ensures that each caller receives prompt and professional service.

General

GENERALANNOUNCEMENT SETTINGSOTHERS

Code *

Description *

Strategy

CID Name Prefix

Join Announcement

Agent Announcement

Service Level

Join Empty

Leave When Empty

Alert Info

Round Robin Memory

None

None

Value in seconds

Yes

No

Queue Ring Time

Member Ring Time

Retry

Wrap-up-time

Music on Hold

Force Music on Hold

Ring Busy Agents

Ring Unavailable Agents

Record

0

15

5

0

Default

No

No

No

No

Members

Extension	Penalty	Member Type	Allow Diversions

Final Destination *

After Agent Hangup Destination

Select Module

Select Destination

Select Module

Select Destination

Save

Code*, number to reach this service.

Description*, short Description to identify this queue.

Strategy, this defines the strategy to ring this queue. Options are:

- **Ring All:** Ring all available channels until one of the members answers the phone.
- **Least Recent:** Ring interface which was least recently hung up by this queue.
- **Fewest Calls:** Ring the one with fewest completed calls from this queue.
- **Random:** Ring random interface.
- **Round Robin Memory:** Round robin with memory, remember where we left off last ring pass.
- **Round Robin Ordered:** Same as Round Robin Memory, except the queue member order from config file is preserved.
- **Linear:** Rings interfaces in the order specified in this queue. If you use dynamic members, the members will be rung in the order in which they were added.
- **Weight Random:** Rings random interface but uses the member's penalty as a weight when calculating their metric.

CID Name Prefix, prefix to append to this queue, typically indicate to the agents from which queue the call comes.

Join Announcement, allowing you to define an announcement to be played to

the caller immediately as they reach the queue.

Agent Announcement, an announcement may be specified which is played for the member as soon as they answer a call, typically to indicate to them which queue this call should be answered as, so that agents or members who are listening to more than one queue can differentiated how they should engage the customer.

Service Level, the idea is to define the maximum acceptable time for a caller to wait before being answered. You then note how many calls are answered within that threshold, and they go toward your service level. So, for example, if your service level is 60 seconds, and 4 out of 5 calls are answered in 60 seconds or less, your service level is 80%.

Join Empty, If there are calls queued, and the last agent logs out, the remaining incoming callers will immediately be removed from the queue, and the Queue () call will return, If "leave when empty" is set to "strict".

"join empty" set to "strict" will keep incoming callers from being placed in queues where there are no agents to take calls. The Queue () application will return, and the dial plan can determine what to do next.

This setting controls whether callers can join a queue with no members. There are four choices:

- **yes** – callers can join a queue with no members or only unavailable members.
 - **no** – callers cannot join a queue with no members.
 - **strict** – callers cannot join a queue with no members or only unavailable members.
- **loose** – same as strict, but paused queue members do not count as unavailable (new in 1.6) **Leave When Empty**, used to control whether callers are kicked out of the queue when members are no longer available to take calls.

Timeout Priority, this is used to control the priority of the two possible timeout options specified for a queue. The Queue (App) application has a timeout value that can be specified to control the absolute time a caller can be in the queue. The timeout value controls the amount of time (along with retry) to ring a member for. Sometime these values conflict, so you can control which value takes precedence.

Queue Timeout, this will cause the queue to fail out after a specified number of seconds.

Member Timeout, this specifies the number of seconds to ring a member's device. **Retry**, specifies the number of seconds to wait before attempting the next member in the queue if the timeout value is exhausted while attempting to ring a member of the queue.

Wrap-up time, the number of seconds to keep a member unavailable in a queue after completing a call. This time allows an agent to finish any post call processing they may need to handle before they are presented with the next call.

Queue Callback, when you have the Queues Callback add-on installed, here you can select the Queue Callback for this Queue.

Music on Hold, this option sets the class of music for this queue.

Ring Busy Agent, this is used to avoid sending calls to members whose status is In Use.

Record, record the calls in this queue.

Members section

Extension, extension number for the Queue Member.

Penalty, within a queue, we can penalize members in order to lower their preference for being called when there are people waiting in a particular queue. For

example, we may penalize queue members when we want them to be a member of a queue, but to be used only when the queue gets full enough that all our preferred agents are unavailable. This means we can have three queues (say, Support, Sales, and Billing), each containing the same three queue members: James Shaw, Kay Madsen, and Danielle Roberts. Suppose, however, that we want James Shaw to be the preferred contact in the Support Queue, Kay Madsen preferred in Sales, and Danielle Roberts preferred in Billing. By penalizing Kay Madsen and Danielle Roberts in Support, we ensure that James Shaw will be the preferred queue member called. Similarly, we can penalize James Shaw and Danielle Roberts in the Sales Queue, so Kay Madsen is preferred, and penalize James Shaw and Kay Madsen in the Billing Queue so Danielle Roberts is preferred.

Member Type, this decides if the member will be **Dynamic** or **Static**.

Allow Diversion, decide if the member executed or not the diversions when called.

Final Destination*

The destination if the call is not answered after the "Queue Timeout".

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

After Agent Hang-up Destination

This option is generally used when performing quality surveys for customer service. Once the call is hung up by the agent, we can send this call to someone to survey the customer or to an IVR tree with all the options. Afterwards, in the case of the IVR tree, we can use the IVR stats add-on to export the information based on the options selected by the customers.

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

Announcement Settings

GENERAL

ANNOUNCEMENT SETTINGS

OTHERS

Periodic Announcements

Periodic Announcement

Default

Periodic Announcement Frequency

Value in seconds

Announce First User

No

Relative Periodic Announcement

Yes

Announce Hold Time

No

Position Announcements

Announce Position

No

Announce Position Limit

Number of zero or greater

Announce Frequency

Value in seconds

Min Announcement Frequency

Value in seconds

Announce Round Seconds

0 Seconds

Save

Periodic Announcement, periodic announcement to provide to the caller. The system default message is "All representatives are currently busy assisting other callers. Please wait for the next available representative."

Periodic Announcement Frequency, this indicates how often we should make periodic announcements to the caller. Bear in mind that playing a message to callers on a regular basis will tend to upset them. Pleasant music will keep your callers far happier than endlessly repeated apologies or advertising, so give some thought to:

- keeping this message short
- Don't play it too frequently.

Announce First User, if enabled, play announcements to the first user waiting in the Queue. This may mean that announcements are played when an agent attempts to connect to the waiting user, which may delay the time before the agent and the user can communicate.

Relative Periodic Announcement, if set to yes, the Periodic Announce Frequency timer will start from when the end of the file being played back is reached, instead of from the beginning.

Announce Hold Time, defines whether the estimated hold time should be played along with the periodic announcements.

Announce Position, defines whether the caller's position in the queue should be announced. If you have any logic in your system that can promote callers in rank (i.e., high-priority calls get moved to the front of the queue), it is best not to use this option. Very few things upset a caller more than hearing that they've been moved toward the back of the line. Options are:

- **No:** The position will never be announced
- **Yes:** The caller's position will always be announced
- **Limit:** The caller will hear her position in the queue only if it is within the limit defined by Announce Position Limit.
- **More:** The caller will hear her position if it is beyond the number defined by Announce Position Limit.

Announce Position Limit, used if you've defined Announce Position as either limit or more.

Announce Frequency, defines how often we should announce the caller's position and/or estimated hold time in the queue. Set this value to zero to disable. In a small call center, it is unlikely that the system will be able to make accurate estimates, and thus callers are more likely to find this information frustrating.

Min Announce Frequency, this specifies the minimum amount of time that must pass before we announce the caller's position in the queue again. This is used when the caller's position may change frequently, to prevent the caller hearing multiple updates in a short period of time.

Announce Round Seconds, if this value is nonzero, the number of seconds is announced and rounded to the value defined.

Others

GENERAL ANNOUNCEMENT SETTINGS OTHERS

Member Settings

Autopause

No

Penalty Members Limit

Number of zero or greater

Member Delay

Value in seconds

Timeout Restart

No

Mark Cancelled Calls as Answered

No

Other Queue Settings

Queue Weight

Value of 0 or higher

Queue Max Length

0

Reset Stats

Disabled

Answer Channel

Yes

Force Longes Waiting Caller

No

IVR

None

VIP Customers

None

Autofill

Yes

Save

Members Settings

Auto Pause, this enables/disables the automatic pausing of agents who fail to answer a call. A value of All causes this agent to be paused in all queues that they are a member of. This parameter can be tricky in a live environment, because if the agent doesn't know they've been paused, you could end up with agents waiting for calls, not knowing they've been paused. Never use this unless you have a way to indicate to the members that they've been paused or have a supervisor who is watching the status of the queue in real time.

Penalty Members Limit, a limit can be set to disregard penalty settings when the queue has too few members. No penalty will be weighed in if there are only X or fewer queue members.

Member Delay, this is used if you want a delay prior to the caller and queue member being connected to each other.

Timeout Restart, if set to yes, resets the timeout for an agent to answer if either a BUSY or CONGESTION status is received from the channel. This can be useful if the agent is allowed to reject or cancel a call.

Other Queue Settings section

Queue Weight, this option defines the weight of a queue. A queue with a higher weight defined will get priority when members are associated with multiple queues. Keep in mind that if you have a very busy queue with a high weight, callers in a lower-weight queue might never get answered (or must wait for a long time).

Queue Max Length, this value specifies the maximum number of callers allowed to be waiting in a queue. A value of zero means an unlimited number of callers are allowed in the queue.

Reset Stats, it allows you to select a Cron Profile to reset the queue stats periodically.

IVR, an IVR may be specified, in which if the user types a SINGLE digit extension while they are in the queue, they will be taken out of the queue and sent to that

extension in this IVR.

VIP Customer, List of VIP Customers, these customers have more priority in this queue.

Autofill, the old behavior of the queue (autofill=no) is to have a serial type of behavior in that the queue will make all waiting callers wait in the queue even if there is more than one available member ready to take calls until the head caller relates to the member they were trying to get to.

5.4.3 Queues Priorities

Here, you can change the weight of a queue dynamically, in order to prioritize calls that incoming for this way. Usually, the destination is a queue.

General

GENERAL

Description *

Priority

10

Destination *

Select Module

Select Destination

Save

Description*, short Description to identify this queue priority.

Priority, the Queue Priority to set.

Destination section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

5.4.4 Queues VIPs

These are the customers who will have a higher priority when calling.

General

GENERAL

Description *

VIP List *

Save

Description*, short description to identify this.

VIP List*, here you can insert the list of numbers separated by line.

5.4.5 Queues Callback

With the Queues Callback module, you can reduce customer frustration by minimizing the time spent on hold. This feature provides callers with the option to request a callback from the next available agent instead of waiting on hold, allowing them to disconnect from the call and tend to other things.

How it Works?

When someone calls your company and there are no available agents, after the customers have waited for a predefined length of time, an automated message can offer to call them back. If the customer decides to request the callback service, their number will be saved and queued. When an agent becomes available, Attimo CloudPBX will then automatically call the person who left the callback request. If that person answers the call, they will be connected to the agent.

Why Use It?

There are many reasons this module can be effective for you, but between those reasons you can find:

Increased customer satisfaction and retention: providing this feature to your customer, you let them know that you value their time. It is a courtesy that boosts the customers' impression of your service and results in higher satisfaction with the customer experience.

Reduce call abandonment rate: When your customers have the option to request a callback, they will no longer be tempted to hang up and move on from your services. Instead, they can request a callback and go about their day while they wait for an agent to return their call.

Manage high volumes: Whether you regularly experience peak periods at your call center or occasionally have spikes in call volume, callbacks can defer calls until volumes are more manageable. "Smoothing out" peak periods make more efficient use of agents, improving call center productivity and reducing the need to hire additional resources.

Lowering Telco costs: When you keep a caller in a queue, a PSTN line is occupied the whole time. Often, a toll-free DID at a premium per-minute rate. The callback feature eliminates the need to keep lines open, removing the telco costs associated with the queue waiting time.

Boost employee morale: Your customers don't like waiting in queues, and your agents don't like dealing with customers who have been kept waiting. When your customers haven't had to wait on hold for a long time to reach an agent, they are more likely to be in a better mood when speaking with your agents. That reduces stress on your agents and helps them improve their productivity.

In the Queues Callback module, you need to configure the following options:

GENERAL

Description *

CallBack Queue

Class of Service

Dial Prefix

Time Group

Maximum Tries

Ring Time

Same Queue

All Permissions

None

3

15

Caller ID

DTMF Digit

Instructions Message

Invalid Message

Number Prompt Message

Thanks Prompt

Ask CallBack Number

Caller ID Name

Caller ID Numt

1

Default

Default

Default

Default

No

Allowed Number Rules

Pattern	Enabled	
	Yes	

Add

Save

Description, brief description.

Callback Queue, it allows you to define the queue where callers who request a callback will be sent.

Class of Service, it allows you to define what Class of Service will be used to perform the callback. Remember that the class of service contains the permissions

to allow outgoing calls through specific trunks or outbound routes.

Dial Prefix, if defined, it will be prepended to the requested callback number.

Time Group, it allows you to configure a time group to define the allowed times to perform callbacks.

Maximum Tries, number of allowed tries to perform before marking the call back as failed.

Caller ID, Caller ID info to use during the callback. This information could be modified during the call process by the trunks or outbound routes settings.

Instructions Message, message to be played with the instructions for requesting a callback.

- **Default Message:** "All of our representatives are currently busy. Please stay on the line and your call will be answered by the next available representative or press one to be called back when a representative is available."

Invalid Message, message to be played when the provided number by caller doesn't match with the defined number rules.

- **Default Message:** "I'm sorry, that is not a recognized phone number"

Number Prompt Message, instructions message for asking to enter the callback number.

- **Default Message:** "Please enter your telephone number"

Ring Time, it allows you to define the time that the callback numbers will ring before marking the call as not answered.

Ask Callback Number, if enabled, the caller will be prompted to enter its callback number. If set to no, the caller id number it will be used as callback number, as long as the number match with the allowed number rules.

Allowed Number Rules, they are the numbers allowed to return the call, Patterns can be used.

Pattern, it allows to define rules of what type of numbers can be used when requesting a call back option:

- X: The letter X or x represents a single digit from 0 to 9.
- Z: The letter Z or z represents any digit from 1 to 9.
- N: The letter N or n represents a single digit from 2 to 9.
- .: wildcard, this matches one or more characters.
- !: wildcard, matches zero or more characters immediately.
- [1237-9]: matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9).
- [a-z]: matches any lower-case letter.
- [A-Z]: matches any Upper-case letter.
- Enabled, it allows you to Enable/Disable pattern rules.

5.5 Calls Routing

5.5.1 Trunks

In the simplest of terms, a trunk is a pathway into or out of a telephone system. A trunk connects Attimo CloudPBX to outside resources, such as PSTN telephone lines or additional PBX systems to perform inter-system transfers. Trunks can be physical, such as a PRI or PSTN line, or they can be virtual by routing calls to another endpoint using Internet Protocol (IP) links.

Trunks are the PBX equivalent of an external phone line. They are the links that allow your system to make calls to the outside world, and to receive calls from the outside world. Without a trunk, you cannot call anyone, and no one can call you. You can configure a trunk to connect with:

Any VoIP service provider.

Any PSTN/Media Gateway, which allows you to make and receive calls over standard telephone lines from your local telephone company.

Connect directly to another PBX.

General

The screenshot displays the 'General' configuration tab for a trunk. The interface is divided into several sections:

- Technology:** Includes fields for Description, Class of Service, Ring Time, Dial Profile, Profile, Music on Hold, Codec, DTMF Mode, and Simultaneous Calls.
- General Configurations:** Includes fields for Local Username, Remote Host, Remote Port, Local Secret, Transport, Contacts, Match, Identify By, Quality Frequency, Quality Timeout, and Max Contacts.
- Outbound Registration Settings:** Includes checkboxes for Require Registration and Permanent Auth Rejection, and fields for Client URI, Server URI, and Contact User.
- Advanced Settings:** Includes fields for Get DID From, Trunk CID, Overwrite CID, Dial Prefix, Disable Trunk, Continue on Busy, Disable Hangup Audios, Calls Recording, Remote Username, Remote Secret, From User, From Domain, Outbound Proxy, Contact Header, Match Header, Trust ID Inbound, Trust ID Outbound, Inband Progress, Max Retries, Expiration, Retry Interval, and Forbidden Retry Interval.

Technology, the type of trunk that you want to create.

- PJSIP.
- SIP.
- IAX2.
- TELEPHONY. (Shown if the DAHDI add-on is installed)
- TENANT. (Shown if a Tenant is created)
- CUSTOM.

SIP, PJSIP and IAX2 trunks utilize the technologies of their namesakes. These trunks have the same highlights and pitfalls that extensions and devices using the same technology do. Telephony trunks require physical hardware cards for incoming lines to plug into. SIP trunks are the most widely adopted and compatible but have difficulties traversing firewalls. IAX2 trunks can traverse most firewalls easily

but are limited to Communications Systems -based systems.

Setting up a trunk is very similar to setting up an extension. All the trunks share common setup fields, followed by fields that are specific to the technology of the trunk.

Description, a description to help identify this trunk.

Class of Service, class of service to be used by this trunk.

Ring Timer, time to ring the trunk before determining that the call cannot be completed.

Dial Profile, there are many options that you can set on the outbound call, including call screening, distinctive ringing, and more. Go to Settings > Technology > Dial Profile for more information.

Profile, profile with common parameters for the technology selected.

Music on Hold, default music on hold for this trunk.

Codecs, list of allowed codecs for SIP trunks, in order of preference. Codecs that are not listed will not be allowed for this trunk. Port, the port number we want to connect to on the remote side.

NAT, (Network Address Translation) is a technology most used by firewalls and routers to allow multiple devices on a LAN with "private" IP addresses to share a single public IP address. A private IP address is an address, which can only be addressed from within the LAN, but not from the Internet outside the LAN. The Options are:

- **No**: Do no special NAT handling other than RFC3581
- **Force**: Pretend there was a report parameter even if there was
- **Comedia**: Send media to the port Communications Systems received it from regardless of where the SDP says to send it.
- **Auto Force**: Set the force report option if Communications Systems detects NAT.
- **Auto Comedia**: Set the comedia option if Communications Systems detects NAT.

Get DID From, it allows you to define from which SIP header will be extracted the DID number.

Get CID From, it allows you to define from which SIP header will be extracted the caller ID info.

Trunk CID, this sets the default caller ID name and number that will be displayed to the called party. The Trunk CID will only be used if **Overwrite CID** field is set to **Yes**. Note that setting the outbound caller ID only works on digital lines (T1/E1/J1/PRI/BRI/SIP/IAX2), not POTS lines. The ability to set outbound caller ID must also be supported by your provider.

Name, a string that can be used to identify calls on this trunk. If this field is left blank, only the Trunk CID number will be sent.

Number, the telephone number that will be displayed by calls on this trunk.

DTMF Mode, set default dtmf-mode for sending DTMF. Default rfc2833|rfc4733, Options:

- **info**: SIP INFO messages (application/dtmf-relay)
- **shortinfo**: SIP INFO messages (application/dtmf)
- **inband**: Inband audio (requires 64 kbit codec -alaw, ulaw)
- **auto**: Use rfc2833|rfc4733 if offered, in-band otherwise.

Overwrite CID, Overwrites the CID sent by the Extension or module. You got the following options:

No, no overwrite will take place and the CID number is preserved. Yes, will

overwrite any CID number sent through this route.

If not Provided, will overwrite the CID information if no External CID is provided.

Disable Trunk, this allows you to disable this trunk to be inaccessible.

Continue on Busy, it forces to continue the call to the next configured trunk when this trunk being busy. **Note:** The call will also continue to the next trunk if any error happens, even if this checkbox is not checked.

SIP/PJSIP/IAX settings

Device for Outgoing Calls (Peer)

Outgoing Username, Username the remote server should use to contact this PBX. It is also the device name that will be created.

Host, this is the IP address or DNS hostname of the SIP provider. This is the destination server or network that Attimo CloudPBX will send calls to when using this trunk.

Port, this value sets the default port to be accessed on the remote endpoint device. Only required for SIP trunks.

Local Secret, secret to be used for authentication requests from remote server.

- **Insecure:** Allows relaxing authentication of incoming SIP requests.
Options:
- **Port:** Allow matching of peer by IP address without matching port number.
- **Invite:** Do not require authentication of incoming 'INVITES'.
- **Port, Invite:** The combination is the minimum security since no checking or port check or authentication to the INVITE message type.

Allow Inbound Calls, if checked, this device will be allowed also to accept calls.

Remote Username, authentication username for remote server.

Remote Secret, the password credential used to authenticate this trunk against the provider.

From User, the user credential used to authenticate this trunk against the provider.

From Domain, as your provider knows your domain.

Qualify, causes Attimo CloudPBX to regularly send a SIP OPTIONS command to check that the peer is still online. If the peer does not answer within the configured period, Attimo CloudPBX will consider the device to be off-line and not available for future calls.

IAX Trunking, allows sending voice of several calls in one IAX packet. It can significantly reduce the required network bandwidth.

Device for Incoming Calls (User)

Username, the username credential used to contact this trunk.

Host, the host they use to contact us (We could specify the "dynamic" option and leave open the possibility that any device connected to your machine without an IP in particular.)

Local Secret, secret to be used for authentication requests from remote server.

Insecure, Sets the level of authentication and verification established between machines when performing communication. Options are:

- **Port:** Allow matching of peer by IP address without matching port number.

- **Invite:** Do not require authentication of incoming INVITEs.
- **Port, Invite:** The combination is the minimum security since no checking or port check or authentication to the INVITE message type.

IP Authentication, if checked, allows the incoming requests authentication by IP address in addition to the username authentication.

Qualify, make periodic checks to make sure that the user is alive.

IAX Trunking, allows sending voice of several calls in one IAX packet. It can significantly reduce the required network bandwidth.

General Configurations (PJSIP)

Transport, explicit transport configuration to use.

Match, the value is a comma-delimited list of IP addresses or hostnames. IP addresses may have a subnet mask appended. The subnet mask may be written in either CIDR or dotted-decimal notation. Separate the IP address and subnet mask with a slash ("/").

Contacts, Permanent contacts assigned to an AoR. You can define multiple contact addresses in SIP URI format. e.g.: sip:198.51.100.1:5060.

Outbound Registration Settings (PJSIP)

Require Registration, it defines, if is required to register against the remote server or VoIP provider.

Permanent Auth Rejection, if this option is enabled and an authentication challenge fails, registration will not be attempted again until the configuration is reloaded.

Client URI, this is the address-of-record for the outbound registration (i.e. the URI in the header of the REGISTER). For registration with an ITSP, the client SIP URI may need to consist of an account name or number and the provider hostname for their register, e.g. 1234567890@example.com. This may differ between providers.

Server URI, this is the URI at which to find the register to send the outbound REGISTER. This URI is used as the request URI of the outbound REGISTER request from Communications Systems.

Contact User, this establishes the user portion from the SIP contact header of the SIP URI. This will affect the reached extension on the Dial Plan when the far end calls you at this registration.

Max Retries, maximum number of registration attempts.

Expiration, expiration time for registrations in seconds.

Retry Interval, interval in seconds between retries if outbound registration is unsuccessful.

Forbidden Retry Interval, it defines the time to wait before attempting registration again, after receiving a 403 Forbidden response. If 0 is specified, no retry will be made after receiving a 403 Forbidden response.

- For registration to generic register, the client SIP URI will depend on networking specifics and configuration of the register.
- For registration with an ITSP, the setting may often be just the domain of the register, e.g. sip:sip.example.com

Register String

Register String, the register line includes a host name (mydomain.com) which tells Communications Systems where to send the registration request; the account number and password, for example: account:password@mydomain.com:5060

5.5.2 Outbound Routes

Outbound Routes enable you to choose which Trunks (phone lines) to use when users dial external telephone numbers. A simple installation will direct the PBX to send all calls to a single trunk. However, a complex setup could have an outbound route for emergency calls, another outbound route for local calls, another for long distance calls, and perhaps even another for international calls.

You can even create a "dead trunk" and route prohibited calls (such as international and premium calls) to it.

Outbound Routes are a set of rules that Attimo CloudPBX uses to determine which trunk to use for an outbound call. Many VoIP systems have access multiple trunks, as it can be unnecessarily expensive to route all calls over a single trunk. Outbound routing also allows dialed numbers to be rewritten on the fly (to remove or prepend dialed numbers with specific outside access codes or area codes). Routes are defined using patterns, against which the dialed numbers are matched.

Outbound routes have a priority. If a dialed number matches the pattern in two outbound routes, the route with the lower priority rating will be used to place the call. The priority is determined when you define an outbound route: The Route Position list in the Route Settings section determines the sequence in which outbound routes are tested, until a match is found.

General

The screenshot shows the 'GENERAL' settings for an outbound route. It includes the following fields and sections:

- Description ***: Text input field.
- Trunks ***: List selection field with a menu icon.
- PIN**: Dropdown menu with 'None' selected.
- CSV**: File upload field.
- Outbound CID**: Fields for 'CID Name' and 'CID Number'.
- Overwrite CID**: Dropdown menu with 'No' selected.
- Intra-Company**: Toggle switch set to 'No'.
- Calls Recording**: Toggle switch set to 'No'.
- Dial Patterns ***: A table with columns: Prepend, Prefix, Pattern, CID Pattern. It includes input fields for each column and an 'Add' button.
- Failover Destination**: Two dropdown menus labeled 'Select Module' and 'Select Destination'.
- Buttons**: 'Download CSV Format' and 'Save' at the bottom.

Description*, short description to identify this outbound route. The name is usually descriptive of the purpose of the route (for example, "local" or "international").

Trunks, list of trunks to use. The order of these matter, the top one will have a higher priority over the others below.

PIN, from PIN Lists, using the previously created password sets (if any), to authenticate access to this route. PIN sets can be configured in the Attimo CloudPBX PIN dialog.

Outbound CID, it allows you to define a specific Caller ID Number/Name that will be using on this outbound route when the "Overwrite CID" setting is enabled.

Overwrite CID, Overwrites the CID sent by the Extension or module. You got the following options.

- **No**, no overwrite will take place and the CID number is preserved.
- **Yes**, will overwrite any CID number sent through this route.

- **If not Provided**, will overwrite the CID information if no External CID is provided.

Intra-Company, if checked, the internal caller id will be sent through this outbound route, instead of the external caller id of the calling extension.

Dial Patterns:

- **Prepend**, digits to add to a successful match.
 - **Prefix**, prefix to remove to a successful match.
 - **Pattern**, pattern matching allows us to create extension patterns in our dial plan that match more than one possible dialed number
- Options:

- X: The letter X or x represents a single digit from 0 to 9.
- Z: The letter Z or z represents any digit from 1 to 9.
- N: The letter N or n represents a single digit from 2 to 9.
- .: wildcard, this matches one or more characters.
- !: wildcard, matches zero or more characters immediately.
- [1237-9]: matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9).
- [a-z]: matches any lower-case letter.
- [A-Z]: matches any Upper-case letter.

CID Pattern, if defined, calls that match with the Dial Pattern, will also need to have a matching CID number to place a call through this route. The CID number to take into consideration is the Internal CID not the External CID. You got the following options.

- X: The letter X or x represents a single digit from 0 to 9.
- Z: The letter Z or z represents any digit from 1 to 9.
- N: The letter N or n represents a single digit from 2 to 9.
- .: wildcard, this matches one or more characters.
- !: wildcard, matches zero or more characters immediately.
- [1237-9]: matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9).
- [a-z]: matches any lower-case letter.
- [A-Z]: matches any Upper-case letter.

Be sure that the outbound routes you create allow you to dial the following types of calls:

Emergency: Dedicate a route just for this purpose. Calls for emergency services should never be mangled by another dial pattern.

Local: Calls to local numbers (usually NXXXXXXXXX).

Toll-free: Calls to toll-free numbers (such as 1-888 or 1-800 numbers).

Mobiles: Ensure that your outbound routes have been configured to handle calls to all mobile phone providers.

International: Calls outside of the country, if permitted (usually 011).

Special: Calls that do not fit any other category. This includes calls such as calls to the operator (0) and directory assistance (411).

Long distance: Calls outside of the local calling area, if permitted (usually 1NXXXXXXXXX). Make sure that your outbound routes are designed to properly handle calls if you are using a dedicated provider for international calls.

5.5.3 Inbound Routes (DID)

The Inbound Routes module is where you define how the PBX handles incoming calls. Typically, you determine the phone number that outside callers have called (DID Number) and then indicate which extension, Ring Group, Voicemail, or other destination to which the call should be directed.

General

GENERAL

Routing Method: Default

Description *

DID Pattern

CID Pattern

Caller ID Modifier: None

CID Lookup: None

Language: Portuguese (Brazil) (pt_BR)

Music on Hold: -- None --

Alert Info

Drop Anonymous Calls: No

Enable Recording: No

Privacy Manager: No

Fax Settings

Fax Detection: No

Inbound Destination *

Select Module

Select Destination

Save

Routing Method, here you can either be the Telephony channel for an analog port (FXO), or default for all other inbound routes like E1, T2, Sip or IAX trunk. From version 2.3.8 onward it is possible to route a DID range to an extension number. For example, if I have the DID range from 1 (305) 6724 7100 to 1 (305) 6724 7200, it is possible to get the last four digits of the DID and route it to the corresponding extension.

Description*, short description to identify this DID. This field is used to hold a description to help you remember what this inbound route is for. This field is not parsed by Attimo CloudPBX.

DID Pattern*, expected number or pattern. This field is used when DID-based routing is desired. The phone number of the DID to be matched should be entered in this field. The DID number must match the format in which the provider is sending the DID. Many providers will send the DID information with the call as +15555555555, while others will leave out the country code information and simply send 5555555555. If the DID entered in this field does not exactly match the number sent by the provider, then the inbound route will not be used. This field can be left blank to match calls from any DIDs (this will also match calls that have no DID information). This field also allows patterns to match a range of numbers. Patterns must begin with an underscore (_) to signify that they are patterns. Within patterns, X will match the numbers 0 through 9, and specific numbers can be matched if they are placed between square parentheses. For example, to match both 555-555-1234 and 555-555- 1235, the pattern would be _555555123[45].

CID Pattern, CID number or CID pattern to match. This can be used when CID-based routing is preferred. As with the DID Number field, the CID entered in this field must exactly match the format in which the provider is sending the CID. Providers

may send 7, 10, or 11 digits; they may include a country code and the plus symbol. Check with your provider to see the format in which the CID is sent, to ensure that this field is entered correctly.

- The Caller ID Number field can be left blank to match with any CIDs (this will also match calls that have no CID information sent with them). The field allows Private, Blocked, Unknown, Restricted, Anonymous, and Unavailable values to be entered, as many providers will send these in the CID number data.

Leaving both the DID Number and Caller ID Number fields blank will create a route that matches all calls.

Caller ID Modifier, this allows you to modify the caller id name/number.

CID Lookup, it allows you to select a CID Lookup item to search the incoming caller number into a directory of a CRM or a cloud directory, or other and set the correct CID Name.

Language, this option specifies the language setting to be used for this route. This will force all prompts specific to the route to be played in the selected language, provided that the language is installed and voice prompts for the specified language exist on your server. This field is not required. If left blank, prompts will be played in the default language of the Attimo CloudPBX server.

Music on Hold, this drop-down menu allows you to select the music-on-hold class for this route. Whenever a caller accessing this route is placed on hold, they will hear the music on hold defined in the class selected here. This is often used for companies that use their music on hold to advertise services, or that accept calls in multiple different languages. Calls to a French DID might play a music-on-hold class with French advertisements, while an English DID would play a class with English advertisements. **Alert Info**, to set a distinctive ring for this inbound route. There does not yet seem to be a standard for how to tell a SIP phone that you want it to ring with a distinctive ring. On SIP handsets that do support distinctive ringing, the exact method of specifying distinctive ring varies from one model to another. In many cases this is done by sending a SIP_ALERT_INFO header, but the usage of this header is not consistent. This is often used for SIP endpoints that can ring differently, or auto-answer calls based on the SIP_ALERT_INFO text that is received. Any inbound call that matches this route will send the text in this field to any SIP device that receives the call.

Enable Recording, enable call recording on this route.

[Privacy Manager section](#)

Privacy Manager, this drop-down menu is used to enable or disable the Attimo CloudPBX privacy manager functionality. When enabled, incoming calls that arrive without an associated caller ID number will be prompted to enter their telephone number. Callers will be given a few attempts (as defined in the Max attempts field, below) to enter this information before their call is disconnected.

[Destination* Section](#)

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

5.5.4 Dynamic Routing (Auto CLIP Routes)

This feature allows you to route missed or not completed outgoing calls to the original caller. When extension user makes an outgoing call, the called party can call back extension user directly, no pass through the Inbound route setting.

General

GENERAL

LIST

Expiration Time

8 Hours

Digits Match

Delete Used Records

Yes

Only Keep Missed Calls

Yes

Save

Expiration Time, it allows to define how long the records of the dynamic routing list will be conserved before they are automatically deleted.

Digits Match, it allows to define the number of digits to be extracted from the caller’s number (from right to left), to later search for matches in the dynamic routing list. If left blank, the entire caller’s number will be used to search for matches.

Delete Used Records, if checked, the record will be deleted from the dynamic routing list when the callback is answered.

Only Keep Missed Calls, if checked, only calls (outgoing) that are missed by the called party will be saved in the dynamic routing list.

List

GENERAL

LIST

Call Date & Time	Extension Number	Called Number	Trunk	Completed	Actions

Save

Displays the complete list of calls that are pending completion of the Dynamic Route action. The action could be canceled through the options of the Actions button.

Note: For the Dynamic Route to work, the option in the Extension configuration in the Advanced TAB must be activated.

5.6 Incoming Calls Tolls

5.6.1 IVR

IVR (Interactive Voice Response) allows you configure an auto attendant to answer calls and redirect the call-in response to input from the caller. An IVR system is often referred to as a digital receptionist. An IVR plays a pre-recorded message to the caller that asks them to press various buttons on their telephone depending on which department or person they would like to speak with. The IVR system will then route the call accordingly.

Attimo CloudPBX's IVR allows any digits to be defined for routing purposes. For example, pressing "1" could route the caller to the sales ring group. Destinations can be defined to receive the call if the IVR times out or does not receive valid input.

It is important that you carefully plan the call flow and branching options for IVRs, while considering the user experience. IVRs use customized Announcements, so you will need to make sure that they are clear and meaningful, and configured to optimize the caller experience. Factors that you should consider include:

Handling the timeout when there is no input from the caller.

Controlling the action to take if caller provides invalid user input. Allowing the caller to backtrack if s/he has made a mistake or gets lost. Allowing the caller to return to the IVR if voicemail is encountered.

Whether or not to take advantage of time-based branching, by defining Time Groups, for normal office hours, that include start and end times, start and end days of the week, and much more.

Defining a Time Condition and setting one destination if the time matches and a different destination if the time does not match.

The screenshot displays the 'General' configuration page for an IVR in the Attimo CloudPBX system. The page is divided into two main sections: 'GENERAL' and 'ENTRIES'. The 'GENERAL' section contains the following fields and options:

- Description ***: A text input field.
- Class of Service**: A dropdown menu with 'Extensions Only' selected.
- Invalid Tries**: A dropdown menu with '3' selected.
- Welcome Message**: A dropdown menu with 'Default' selected and a speaker icon.
- Instructions Message**: A dropdown menu with 'None' selected and a speaker icon.
- Invalid Retry Message**: A dropdown menu with 'Default' selected and a speaker icon.
- Invalid Message**: A dropdown menu with 'Default' selected and a speaker icon.
- Timeout**: A text input field with '10' entered.
- Invalid Destination ***: Two dropdown menus labeled 'Select Module' and 'Select Destination'.
- Timeout Destination ***: Two dropdown menus labeled 'Select Module' and 'Select Destination'.
- Timeout Tries**: A dropdown menu with '3' selected.
- Timeout Retry Message**: A dropdown menu with 'Default' selected and a speaker icon.
- Timeout Message**: A dropdown menu with 'Default' selected and a speaker icon.
- Welcome after Timeout**: A button labeled 'Yes'.
- Welcome after Retry**: A button labeled 'Yes'.
- Direct Dial**: A button labeled 'No'.
- Generate Stats**: A button labeled 'No'.

A green 'Save' button is located at the bottom right of the configuration area.

Description*, short description to identify this IVR. This field is not parsed by Attimo CloudPBX.

Class of Service, here you can choose a Class of Service for this IVR.

Invalid Tries, number of invalid attempts allowed.

Welcome Message, welcome message, selected from a drop-down menu of pre-recorded messages that will be played to the caller when they enter the IVR.

Instructions Message, Message to be played after the welcome message. This message is useful for avoid repeating the welcome message on invalid/timeout

event. **Invalid Retry Message**, message, selected from a drop-down menu of pre-recorded messages, to be played when the IVR receives an invalid option.

Invalid Message, message, selected from a drop-down menu of pre-recorded messages, to be played when user exceeds the maximum number of attempts.

Timeout, this value is the maximum time, in seconds, that the system will wait for input from the caller. If this time passes without input, the call will fail over to the Timeout Destination that the user has defined.

Timeout Tries, is used to determine the number of times the IVR will repeat itself when no valid input is received. After the specified number of tries, the caller will be sent to the Timeout Destination. The maximum number of loops allowed is five.

Timeout Retry Message, message, selected from a drop-down menu of pre-recorded messages, to be played when input has not been received within the period defined by Timeout. If the number of Timeout Tries has not yet been reached, then the user will be prompted to try again.

Timeout Message, message selected from a drop-down menu of pre-recorded messages, to be played when reaching the timeout.

Welcome after Timeout, when enabled, will return the user to the main IVR Welcome Message after playing the Timeout Retry Message.

Welcome after Retry, when enabled, will return the user to the main IVR Welcome Message after playing the Invalid Retry Message.

Direct Dial, this enables the caller to dial an extension directly from the IVR. If this option is not enabled, the caller will receive a message that they have provided invalid input when they enter an extension, even if the extension is valid.

Generate Stats, if is set on yes, it will be saved stats for each option marked. These statistics can be consulted in the IVR Stats module.

Invalid Destination* Section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

Timeout Destination* Section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

IVR Entries

This tab defines how to handle the user's input.

GENERAL ENTRIES

Digit	Module / Destination	Enabled
Digit to Press	Select Module	Select Destination
	Yes	Add

Save

Digit, digit to press.

Module, module to activate when the caller presses the appropriate digit.

Destination, destination to call when the caller presses the appropriate digit.

Enabled, enabled or disable this option.

It is good practice to ensure that the user has an easy way of getting back to the previous menu. One simple way to do this would be to allow the user to press "*" and link that keypress to the parent IVR.

5.6.2 Time Groups

Time conditions are a set of rules for hours, dates, or days of the week. A condition has two call targets each time. Calls sent to a time condition will be sent to one target if the time of the call matches one of the conditions, or to the other target if none of the conditions match. Each time condition can have multiple time definitions (known as time groups). Time conditions are often used to control how a phone system responds to callers inside and outside of business hours, and during holidays.

Before we can set up a time condition call target, we need to define a set of time groups. Time groups are a list of rules against which incoming calls are checked. The rules specify a specific date or time, and a call can be routed differently if the time it comes in matches with one of the rules in a time group. Each time group can have an unlimited number of rules defined. It is useful to group similar sets of time rules together. For example, there may be one-time group for business hours in which the time that the business will be open will be defined. Another popular time group is for holidays, in which each holiday that falls on a business day is defined.

General

The screenshot shows the 'GENERAL' configuration page for a time group. At the top, there's a 'Description' field with an asterisk indicating it's required. Below this is the 'Schedules' section, which contains a table-like structure with columns for defining time ranges: 'Time to Start', 'Week Day Start', 'Month Day Start', 'Month Start', 'Time to Finish', 'Week Day Finish', 'Month Day Finish', and 'Month Finish'. Each column has a corresponding input field or dropdown menu. A plus icon is on the left to add new rows, and a red minus icon is on the right. An 'Add' button is at the bottom right of the schedule section. At the very bottom right of the page, there is a 'Save' button.

Description*, used to identify the time group, when selecting it during the setup of a time condition. This value is not parsed by Attimo CloudPBX.

Schedules Section

Time to Start, time, in hours and minutes, that the time group should start.

Weekday Start, day of the week that the time group should start.

Month Day Start, day of the month that the time group should start.

Month Start, month of the year that the time group should start.

Time to finish, time, in hours and minutes, that the time group should end.

Weekday Finish, day of the week that the time group should end. **Month Day**

Finish, day of the month that the time group should end. **Month Finish**, month of the year that the time group should end.

5.6.3 Time Conditions

Once a time group has been defined, a time condition can be set up as a call target.

General

TIME CONDITIONS

Toggle Code *

Description *

Time Group *

Time Zone

Authorization Pin

Status

BLF Inverted

Destination if time matches *

Destination if time does not match *

Toggle Code*, dial code to toggle the time condition state through the phone.

Description*, short Description to identify this Time Condition.

Time Group*, select a Time Group, from the drop-down list, that was created in the Time Groups dialog.

Time Zone, as an extended feature, you can set at which Time Zone the Time Condition will be running at.

Authorization Pin, optional password to protect from unauthorized people of modifying this time condition.

Status, allows you to override the default behavior of a time condition, Options:

- **Default:** Default behavior.
- **Temporary Matched/Unmatched:** Overrides temporary the time condition and sends the calls to the matched/unmatched destination until the current time span has elapsed. After that, the behavior will return to default.
- **Permanently Matched/Unmatched:** Overrides permanently the time condition and sends the calls to the matched/unmatched destination until the override is removed.

BLF Inverted, by default the BLF light color is green when the time condition is matching and red when is not matching. Setting up this to "yes" will make that the behavior be the inverse of what is described above.

Destination if Time Matches Section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

Destination if Time does not Match Section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

5.6.4 Announcements

This module is used for when you want the caller to hear a message, before being automatically transferred to a fixed destination.

General

GENERAL

Description *

Custom Recording *

Destination after playing announcement *

Select Module

Select Destination

Save

Description*, short description to identify this Announcement.

Custom Recording, here you can select a recording, from the drop-down list, to play in this Announcement.

Destination after playing announcement* Section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

5.6.5 Languages

This module is used for when you want to change the language of the voice guide in the course of a call, such as when an IVR is used, and the user selects a language.

General

GENERAL

Name *

Language

English (en)

Description *

Destination *

Select Module

Select Destination

Save

Name*, short name, must be unique.

Description, short description to identify this language.

Language, channel language to use from drop-down list.

Destination section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

5.6.6 Night Mode

Night mode is used to change the conditions of an incoming route depending on whether it is active or not.

General

The screenshot shows the 'GENERAL' configuration tab for Night Mode. It includes a sidebar menu icon in the top right. The form contains the following fields and controls:

- Toggle Code ***: A text input field.
- Description ***: A text input field.
- Optional Password**: A text input field.
- State**: A button labeled 'No'.
- Ignore global mode**: A button labeled 'No'.
- Generate Hint**: A button labeled 'No'.
- Destination when Disabled ***: A section containing two dropdown menus: 'Select Module' and 'Select Destination'.
- Destination when Enabled ***: A section containing two dropdown menus: 'Select Module' and 'Select Destination'.
- Save**: A green button with a save icon, located at the bottom right of the form.

Toggle Code*, code to dial to change the night mode state via phone.

Description, short description to identify this Night Mode **Optional Password**, optional password to protect this Night Mode **State**, this indicates whether this Night Mode is currently active.

Ignore global mode, this means that it will not be affected by the state of global night mode.

Generate Hint (indicator), generates hint for this night mode to be seen from a console or monitoring key.

Destination on Destination when Disabled* Section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

Destination on Destination when Enabled* Section

Select Module, allows the user to choose from a drop-down list of available modules, which module should be activated.

Select Destination, this is the call target to which the module should be routed.

5.6.7 CID Modifiers

CID Modifiers allow you to modify the incoming CID name and number: You can append or prepend anything to the CID name/number.

You can replace completely the CID name.

You can cut or delete certain part of the code number.

GENERAL

Description *

Source *

CID Number Settings

Skip / Length

Prepend

Append

CID Name Settings

Prepend

Append

Replace With

Simulation

Original CID Number

Original CID Name

Modified CID Number

Modified CID Name

 Save

Description, short description to identify this CID Modifiers.

CID Number Settings

Skip/Length, this allows you to modify incoming CID number by starting the manipulation several digits either from the beginning or end of the CID number, while retaining any number of the original digits. Options:

- **Skip:** Specify where to start modifying the CID number. A positive skip value of x will ignore the leading x digits. A negative value of x will start x digits before the end of incoming CID number.
- **Length:** determines the length of the modified CID number. If length is zero, all digits after the start position will be used. Define a negative length of x to discard the x trailing digits.

Prepend, prefix to be added at the beginning of the original Caller ID number.

Append, suffix to be added after the original Caller ID number.

CID Name Settings

Prepend, Text that can be added in front of the original Caller ID name.

Append, Text that can be added at the end of the original Caller ID name.

Replace With, completely replace the CID name with this text. Leave this field blank to keep the original CID name.

5.6.8 CID Lookup

With this module it is possible to consult a database or a URL with a telephone number to document the call with the name of the telephone number owner.

General

GENERAL

Description *		Path	
Source *	HTTP/HTTPS	Query String	
Host *		Timeout	
Port		Secure	<input type="radio" value="No"/>
Auth User			
Auth Password			

Save

Description, short description to identify this CID Lookup.

Source, it defines the method to get the CID name of an incoming caller.

Host, it defines the API host to make the request.

Port, it defines the port to make the request. By default, 80 for HTTP request and 443 for HTTPS request.

Auth User, it defines the user to authenticate the HTTP/HTTPS request.

Auth Password, it defines the password to authenticate the HTTP/HTTPS request

Path, it defines the script file name to execute on request. Example: cid_lookup.php

Query String, it defines the arguments needed for execute the script. The special argument value [CIDNUM] it will be replaced by caller cid number. Example: caller_num=[CIDNUM]&ctype=vip.

Secure, if it is checked, the request will be through HTTPS, in other way it will be through HTTP.

5.7 Tools

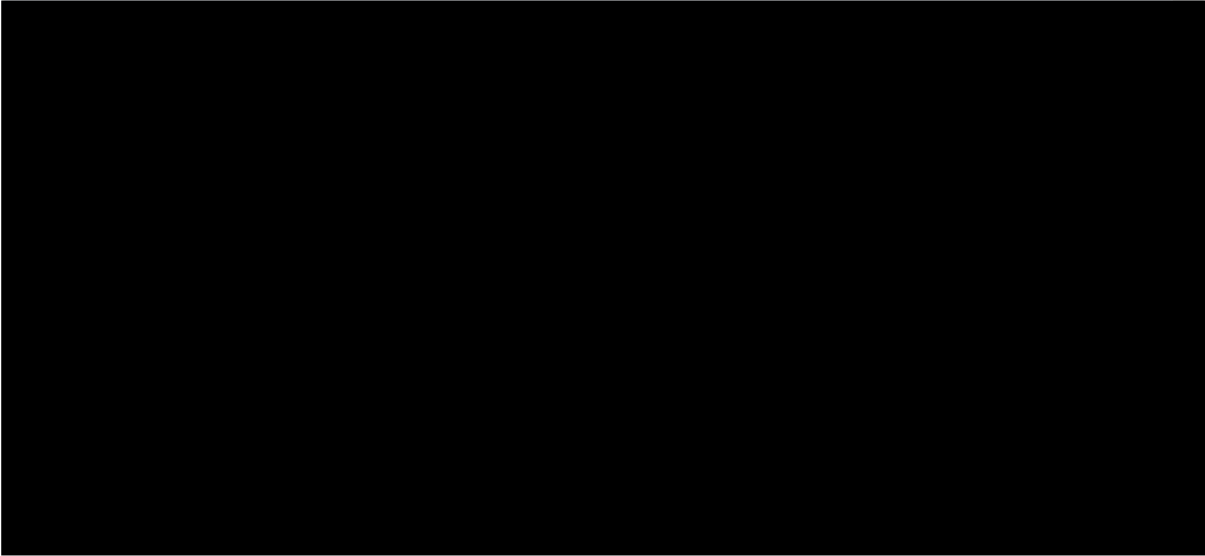
5.7.1 PBX CLI

From this tab you can access the CLI Interface.

General

GENERAL

Type command here!



In this dialog, you can type any valid Communications Systems command. As soon as you start typing, a drop-down list of available Communications Systems commands will be displayed.

5.7.2 Blacklist

Is possible to define a blacklist number with a pattern and define a destination for it. Also, it is possible to disable/enable a blacklist item through the GUI. If no destination is defined for the blacklisted item a message will be played to the caller.

General

GENERAL

CID Number *

Enabled

Yes

Description *

Destination

Select Module

Select Destination

Save

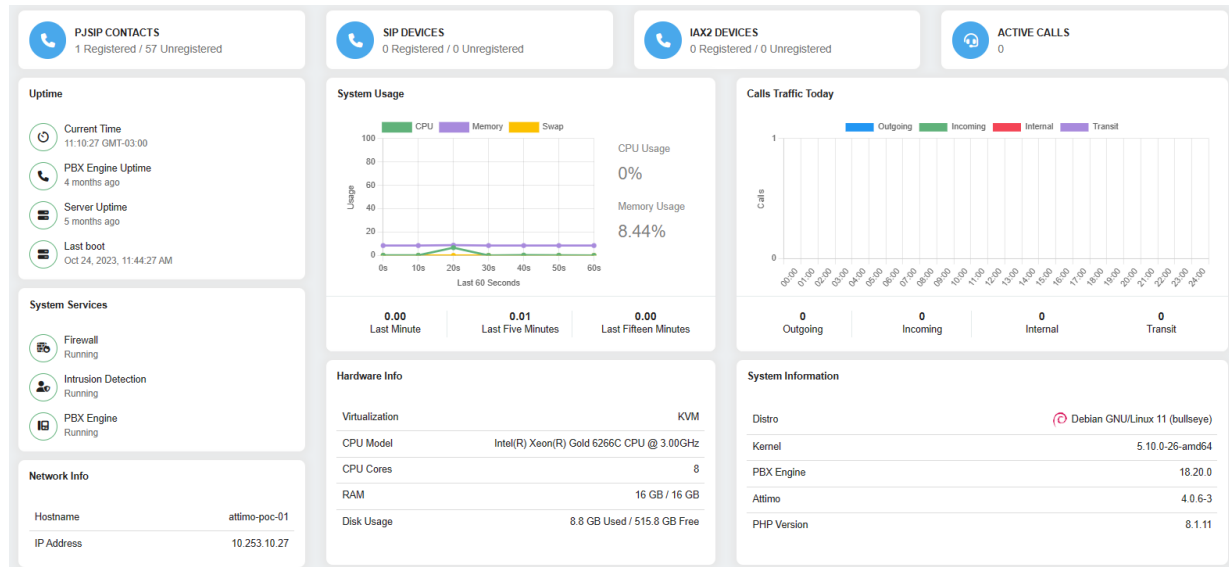
CID Number, the number that you want to blacklist.

Description, a description of the number that you want to blacklist.

Enabled, Enable/Disable this CID number from the blacklist.

Destination, optional destination. If set, the caller with a CID Number that match with the configured above, will be redirected to the configured destination. If not set, a message will be played instead.

5.7.3 Dashboard



From the Dashboard it is possible to monitor various aspects from your PBX. The Dashboard itself is composed of multiple widgets with this information.

Monitor Devices, you can see how many devices are registered for PJSIP, SIP, and IAX2.

Active Calls, here you can view the number of active calls at that moment.

Disk Usage, with this widget you can monitor the usage you have for your storage.

System Usage, this allows you to monitor your CPU, Memory, and SWAP activity.

Call Traffic Today, this widget, available from Version 3 and onward, allows you to track the number of calls you have had for this day. You can monitor the following types of calls.

- **Outgoing**, calls flowing outbound from the PBX.
- **Incoming**, calls flowing inbound from the PBX.
- **Internal**, calls between internal extensions and modules.
- **Transit**, Passthrough calls going through the PBX.

Uptime, this will show the uptime for Communications Systems and the Server. It will also show the current time, and when the server was last booted on.

System Services, on this widget you will see the status of the Firewall, Intrusion Detection (Fail2ban), Communications Systems, and DAHDI. DAHDI only being available if you have installed the DAHDI add-on module.

Hardware Info, this will list the hardware where you have Attimo CloudPBX installed on. This will be the Motherboard or Virtualization software, CPU Model and frequency, CPU Cores, and RAM amount.

System Information, here you will see the information about the current version of various layers of the software. This would be the Distro version, Kernel version, Communications Systems version, and Attimo CloudPBX version.

5.7.4 Log File Viewer

In this tab you will find a tool to view log files.

General

GENERAL

Log File Filter

Lines

[Refresh](#)

Log File, here you can select the log file to see.

Lines, last lines to read from the log file.

Filter, Filter the Log for specific information.

5.7.5 Cron Profiles

In this module the profiles of the tasks to be executed are created.

General

GENERAL

Description Template

Settings

Minute		-- Quick Template --
Hour		-- Quick Template --
Day of Month		-- Quick Template --
Month		-- Quick Template --
Day of Week		-- Quick Template --

[Save](#)

Description, a brief description to identify this profile of Cron.

Template, predefined templates to configure your CRON profile settings easy

and faster.

Settings

Minute, this controls what minute of the hour the command will run on and is between 0 and 59.

Hour, this controls what hour the command will run on, and is specified in the 24-hour clock, values must be between 0 and 23 (0 is midnight).

Day of Month, this is the Day of Month, that you want the command run on, e.g. to run a command on the 19th of each month, the day would be 19.

Month, this is the month a specified command will run on, it may be specified numerically (0-12), or as the name of the month (e.g. May).

Day of Week, this is the Day of Week that you want a command to be run on, it can also be numeric (0-7) or as the name of the day (e.g. sun).

5.7.6 Weak Passwords

Creates a report of any user accounts, extensions, or trunks that have weak registration passwords.

Accounts with weak passwords represent a security hole and should be updated as soon as possible.

In this tab you will find a list of all weak password in your system

General

GENERAL

Show

10

 entries

Search:

Extension	Device	Password	Level
1000 - TESTE	1000 - TESTE	3Corp@POC	Weak

Showing 1 to 1 of 1 entries

Previous1Next

5.8 Extras

5.8.1 Video Conference

Through WebRTC technology, Attimo CloudPBX includes a module which can make your video conferences using our powerful meetings server in the cloud, which have the following options:

Simple to use, no downloads required. VitXi Meet works directly within your browser. Simply share your conference URL with others to get started.

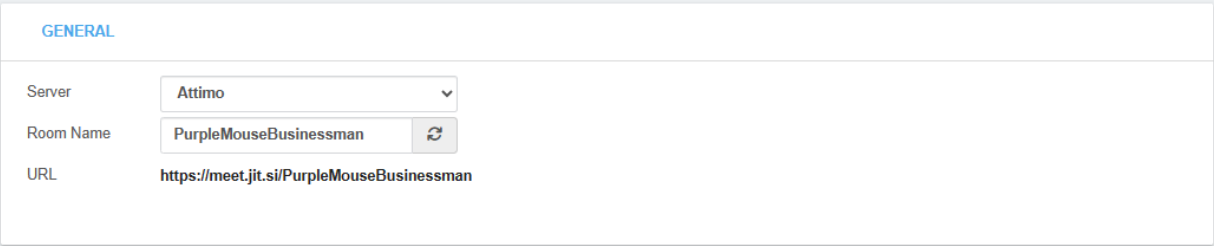
Low Bandwidth, multi-party video conferences work with as little as 128Kbps. Screen- sharing and audio-only conferences are possible with far less.

Unlimited Users, there are no artificial restrictions on the number of users or conference participants. Server power and bandwidth are the only limiting factors.

Screen Sharing, it's easy to share your screen with others. VitXi Meet is ideal for on- line presentations, lectures, and tech support sessions.

Secure Rooms, need some privacy? VitXi Meet conference rooms can be secured with a password to exclude unwanted guests and prevent interruptions.

Share Notes, with VitXi Meet features Etherpad, a real-time collaborative text editor that's great for meeting minutes, writing articles, and more.



The screenshot shows a web form titled "GENERAL" for configuring a video conference. It contains three fields: "Server" with a dropdown menu set to "Attimo", "Room Name" with a text input containing "PurpleMouseBusinessman" and a refresh icon, and "URL" which displays the generated link "https://meet.jit.si/PurpleMouseBusinessman".

Join / Create

Server, here you can select the server where conference will be hosted.

Room Name, a room name to join or start a new conference.

URL, you can share this URL with the other participants of the video conference.

Note: Remember to download our plugins in the Chrome web store, look for them with the word VitXi Meet.

5.9 Emergency Calls

With the introduction of Kari’s Law in the United States, from version 3 and onward, the Emergency module was created. With it, we have two modules, Emergency Numbers and Dispatchable Locations. These two come into play whenever an Emergency Call is placed into action from any extension or device within the PBX.

5.9.1 Emergency Numbers

This module defines the external numbers that cannot be restricted, these numbers can be dialed from any extension, either a Hotdesking that is not registered or any extension that has completely restricted outgoing calls.

General

GENERAL

Description

Email Addresses

Trunks

List of Emergency Numbers

Number	Service Name	
eg.: 911	eg.: Police, Firefighters, Ambulance, Etc.	
		Add

Save

Description, a short description to easy recognize this list of emergency numbers.
Trunks, list of trunks than can be used in case of an emergency call.
Email Addresses, these are the email addresses that are used for the emergency contacts. These emails will have a notification sent to when.
List of Emergency Numbers, telephone number to contact local emergency services for assistance and name of the service that represents this telephone number. e.g.: Police, Firefighters, Ambulance, etc.

5.9.2 Dispatchable Locations

Complying as well with Kari's Law, when have now included this module from Version 3 and onward. This module allows you to create detailed locations which will be used when an Emergency Call is placed. This is the location that is sent to the Emergency Contacts.

General

GENERAL

Name *

Country *

-- Select Country --

Street Address *

Street Number

Street Name

Address Type

-- Select T... --

City *

State *

Postal/ZIP Code *

Caller ID

CID Name

CID Number

Default

No

Save

Name, a brief description to recognize this Dispatchable Location.

Country, the country for this Dispatchable Location.

Street Address, this is the street for this Dispatchable Location.

Address type, the type of address for this location.

City, the city for the location.

State, State, Province or Region.

Postal/ZIP Code, ZIP Code to be used for the location.

Caller ID, Caller ID to be used for this location. In case the Emergency CID for the device or extension is not used, this is the CID used.

Default, if enabled set the location as the default Dispatchable Location for new extensions.

6. Reports

6.1 CDR Reports

6.1.1 CDR Filters

General

The screenshot shows the 'GENERAL' tab of the CDR Filters configuration. At the top, there are input fields for 'Description *', 'Duration' (with 'From' and 'To' sub-fields), and 'Talk Time' (with 'From' and 'To' sub-fields). Below these is a section titled 'Add Search Condition *' which contains a table with columns: Condition, Search By, Value, Exclude, and Mode. The table has one row with 'AND' in the Condition column, 'Caller ID' in the Search By column, an empty Value field, 'No' in the Exclude column, and 'Begins With' in the Mode column. There is an 'Add' button to the right of the table and a 'Save' button at the bottom right of the form.

Condition	Search By	Value	Exclude	Mode
AND	Caller ID		No	Begins With

Description*, short description for this filter.

Duration, call duration range in seconds.

Talk Time, talk time range in seconds.

Add Search Condition Section

Clicking on the Add button allows you add additional search conditions.

Condition, this determines whether the condition is enabled or not. AND → it means that this value must be met in conjunction with the previous one and so on, OR → means that this value or the previous one must be met.

Search By, search for the selected field using one of the items selected from the drop- down list:

- Caller ID.
- Source.
- Destination.
- Account Code.
- Status.
- Customer Code.

Value, value of the selected field.

Exclude, include or exclude the selected value in the search.

Mode, search condition can be filtered in several methods, which can be selected from the drop-down list:

- Begins With.
- Contains.
- Ends With.
- Exactly.

6.1.2 View CDR Reports

General

GENERAL

Filter

None

Source

Destination

From

2024-03-15 00:00:00

To

2024-03-15 23:59:59

Call Records

CSV

PDF

Show 10 entries

Showing 0 to 0 of 0 entries

Previous

Next

Refresh

Filters section

Filter, filter defined in Report Filters dialog, to be used in the report.

Source, call source.

Destination, dialed number.

From, include records starting from this date and time.

To, include records ending by this date and time.

Reports section

Record Per Page, set the number of records that should be displayed on each page.

PDF, CSV, export the report in PDF or CSV.

CEL Event

CEL Events allow you trace the call's steps from the moment it entered the PBX, to when the call was tended to.

CEL Events							
Event Time	Event Type	CID Name	CID Number	Dialed Number	Extension	Context	Application
2020-05-09 20:11:08	CHAN_START	Rodrigo Cuadra	2004		2000	cos-all	
2020-05-09 20:11:08	APP_START	Rodrigo Cuadra	2004	2000	2000	sub-local-dialing	Dial
2020-05-09 20:11:08	CHAN_START	Maynor Peraita	2000		s	cos-all	
2020-05-09 20:11:19	ANSWER	Maynor Peraita	2000		2000	cos-all	AppDial
2020-05-09 20:11:19	ANSWER	Rodrigo Cuadra	2004	2000	2000	sub-local-dialing	Dial
2020-05-09 20:11:19	BRIDGE_ENTER	Maynor Peraita	2000			cos-all	AppDial
2020-05-09 20:11:19	BRIDGE_ENTER	Rodrigo Cuadra	2004	2000	2000	sub-local-dialing	Dial
2020-05-09 20:12:24	BLINDTRANSFER	Rodrigo Cuadra	2004	2000	2000	sub-local-dialing	Dial
2020-05-09 20:12:24	BRIDGE_EXIT	Rodrigo Cuadra	2004	2000	2000	sub-local-dialing	Dial
2020-05-09 20:12:24	HANGUP	Rodrigo Cuadra	2004	2000	h	sub-local-dialing	
2020-05-09 20:12:24	CHAN_END	Rodrigo Cuadra	2004	2000	h	sub-local-dialing	
2020-05-09 20:12:24	BRIDGE_EXIT	Maynor Peraita	2000		2005	cos-all	AppDial
2020-05-09 20:12:24	APP_START	Maynor Peraita	2000		2005	sub-local-dialing	Dial
2020-05-09 20:12:24	APP_START	Maynor Peraita	2000		s	sub-vm	VoiceMail
2020-05-09 20:13:00	HANGUP	Maynor Peraita	2000		h	sub-local-dialing	AppDial
2020-05-09 20:13:00	CHAN_END	Maynor Peraita	2000		h	sub-local-dialing	AppDial

close

6.2 PBX Reports

6.2.1 Active Calls

Shows the ongoing calls in real-time with the following details:

CALLS

Show

100

 entries

Search:

Channel	Caller ID	Called Number	State	Duration	LinkedID
No data available in table					

Showing 0 to 0 of 0 entries

Previous

Next

Channel, channel through which the call is taking place.

Caller ID, CID of the person that placed the call. **Called Number**, Number of the person being called. **State**, current state of the call.

Duration, running duration for the call.

Linked ID, Unique ID that links two channels to the same call.

6.2.2 PJSIP Devices

Shows the status of the endpoints with the following information:

[ENDPOINTS](#) [TRUNKS](#) [OUTBOUND REGISTRATIONS](#)

Show	10	entries	Search:	
Tenant	Extension	Endpoint	Contacts	Max Contacts
Attimo	1000 - TESTE	1000		1
La Pastina	1000 - Aparelho SIP	t2_1000		5
La Pastina	1001 - Softphone 1	t2_1001		1
La Pastina	1002 - Softphone 2	t2_1002		5
La Pastina	1003 - Softphone 3	t2_1003		5
Attimo	1001 - 1001	1001		5
POC Attimo SEAD-GO	1000 - FlyingViceP10G/Khomp K2	t1_1000		5
POC Attimo SEAD-GO	1001 - Yealink T46U	t1_1001		5
POC Attimo SEAD-GO	1002 - Yealink W73P/Grandstream DP 752/DP 722	t1_1002		5
POC Attimo SEAD-GO	1003 - Yealink W73P/Grandstream DP 752/DP 722	t1_1003		1
Showing 1 to 10 of 18 entries				
			Previous	1 2 Next

Extension, Extension number and name.

Endpoint, Endpoint user.

Contacts, Registered contact with its origin and port.

Max Contacts, Max number of contacts for this user.

We can also see the PJSIP Trunks with the following information:

Trunk Description, description for the PJSIP Trunk to monitor.

Endpoint, endpoint user.

Contacts, the different remote contacts that are connected to this trunk.

Match, this value is a list of IP addresses or Hostnames separated by commas. The IP Addresses can have a subnet mask attached. The subnet mask can be written in CIDR or octets.

We can then see the Outbound Registrations with the following information:

Trunk Name, description of the PJSIP trunk to monitor.

Endpoint, endpoint user.

Client URI, this is the registration address for the outbound registration, meaning, the URI on the REGISTER header. To register with an ITSP, the SIP Client URI can be formed by an account name or number and the hostname for the provider, for example, 1234567890@example.com. This may vary between providers.

Server URI, this is the URI to find the register and send the outgoing REGISTER request. This URI is used as the request URI for the outgoing REGISTER request from Communications Systems. For the registration of the ITSP, the configuration can be the register’s domain, for example, sip:sip.example.com.

Status, status for the connection.

6.2.3 SIP Devices

Shows the status for the SIP Devices with the following information:

DEVICES

TRUNKS

REGISTRATIONS

Show

10

entries

Search:

Tenant	Extension	Device	Host	Port	Status	User Agent
No data available in table						

Showing 0 to 0 of 0 entries

Previous

Next

Extension, Extension number and name.

Device, user device for the extension.

Host, from where the user is registered from.

Port, port used to register the user device.

Status, user status.

User Agent, User agent sent by the user. With this information we can know the make and model from the phone the user is registered from.

Then, we can see the Available SIP Trunks with the following information:

Trunk Description, brief description of the trunk to monitor.

Device, device connected to this account.

Host, IP Address from where the registration comes from.

Port, port used for the registration.

Status, device status. This indicates the response time.

Finally, we can see the different registrations with the following information:

Device, device connected to this account.

Host, IP Address from where the registration is coming from.

Port, port used for the registration.

Refresh, how often there is a registration request.

Status, device status. Indicated the response time.

6.2.4 IAX2 Devices

Shows the status of the different IAX2 Devices with the following information:

DEVICES

TRUNKS

REGISTRATIONS

Show

10

▼

entries

Search:

Tenant	Extension	Device	Host	Port	Status
No data available in table					

Showing 0 to 0 of 0 entries

Previous

Next

- Extension,** Extension number and name.
- Device,** user device for the extension.
- Host,** from where the user is registered from.
- Port,** port used to register the user device.
- Status,** user status.

Then, we can see the Available IAX2 Trunks with the following information:

- Trunk Description,** brief description of the trunk to monitor.
- Device,** device connected to this account.
- Host,** IP Address from where the registration comes from.
- Port,** port used for the registration.
- Status,** device status. This indicates the response time.

Afterwards, we can see the different registrations with the following information:

- Device,** device connected to this account.
- Host,** IP Address from where the registration is coming from.
- Port,** port used for the registration.
- Refresh,** how often there is a registration request.

- Status,** device status. Indicated the response time.
- Description,** brief description of the Device to monitor.
- Device,** device connected to this account.
- Host,** IP Address from where the registration comes from.
- Port,** port used for the registration.
- Status,** device status. This indicates the response time.

6.3 IVR Stats

6.3.1 IVR Stats

When we install the IVR statistics module, this option will appear in the Reports menu.

With this module it is possible to obtain statistics of how many times each option is pressed in the IVR, this is very useful to determine which options are not being used in the IVR.

Another application is that of surveys, since this module in combination with the call queue allows us to carry out satisfaction surveys in our Call Center:

GENERAL

IVR

Poc

▼

End Date

2024-03-15

Start Date

2024-03-15

Summary

CSV

Option	Pressed Times
No data available in table	

Detailed

CSV

Call Date	Option	Caller	Callee	Call Duration	Call Status
No data available in table					

Showing 0 to 0 of 0 entries

Previous

Next

Q Search

It also gives us the facility of being able to export the data to CSV.

6.4 Call Center Reports

6.4.1 Queues Call Back Reports

When we install the Queues Call Back add-on module, we will now have this option on the Reports Menu.

On this report we will see the calls that are queued to be called back. These calls opted for this feature while waiting in Queue on our Call Center.

REPORT

Queue CallBack

All

Status

Queued

In Progress

Successful

Failed

Start Date

2024-03-14 00:00:00

End Date

2024-03-15 23:59:59

Scheduled Queue Callbacks

Search:

Date & Time	CallBack Queue	CallBack Requester Name	CallBack Number	Tries	Status	Actions
No data available in table						

Showing 0 to 0 of 0 entries

Previous

Next

Refresh

Here you can see the following information:

Queue Callback, here you can select the Queue Callback you wish to monitor.

Status, here you select the status you want to filter from.

Start Date, from when you wish to see the report.

End Date, when you want the report to end.

Scheduled Queue Callbacks Section

These are all the Callbacks scheduled based on the filters input above.

Date & Time, this is the date and time the Callback was Scheduled.

Callback Queue, the queue where the callback will be sent to. **Callback Requester Name**, the CID name for the Callback requester. **Callback Number**, the number to which the callback will be made to.

Tries, the number of tries the PBX has made to reach the callback number.

Status, the status for the callback.

Actions, actions you can do to this callback. In this case, delete it.

7. Settings

7.1 Technology Settings

7.1.1 PJSIP Settings

The PJSIP Settings is used to configure the default value to be used for PJSIP calls.

General

GENERAL

Debug	No	Unidentified Request Count	3
Bind	0.0.0.0 50751	Unidentified Request Prune Interval	30
TLS Bind	0.0.0.0 50752	Timer T1	500
Tone Zone	(br) Brazil	Timer B	32000
Codecs	ulaw,alaw,gsm,g729,opus,vp8,h264,h263p,h263	Certificate	My Certificate
Language	Portuguese (Brazil) (pt_BR)	SSL Method	tlsv1_2
Keep Alive Interval	90	Disable TCP Switch	Yes
Expiration Check Interval	30	Verify Client	No
Max Initial Quality Time	0	Verify Server	No
Endpoint Identifier Order	Auth Username	Allow Wildcard Certs	No
Unidentified Request Period	5	Allow Transports Reload	Yes
Nat Settings		MWI Settings	
External Media Address	119.8.85.137	Taskprocessor Queue High	500
External Signal Address	119.8.85.137	Taskprocessor Queue Low	-1
Local Net	x 10.253.10.20/32 x 10.253.10.27/32	Disable Initial MWI	Yes

Debug, Enable/Disable SIP debug logging.

Bind, IP Address and optional port to bind to for this transport.

TLS Bind, IP Address and optional port over TLS protocol to bind to for this transport.

Certificate, path to certificate file to present to peer. SSL Method, method of SSL transport (TLS ONLY).

Timer T1, timer T1 is the base for determining how long to wait before retransmitting requests that receive no response when using an unreliable transport (e.g. UDP). Note: Because this value is system, it will only be applied when the Communications Systems service is restarted.

Timer B, timer B determines the maximum amount of time to wait after sending an INVITE request before terminating the transaction. Note: Because this value is a system value, it will only be applied when the Communications Systems service is restarted.

Disable TCP Switch, disable automatic switching from UDP to TCP transports if outgoing request is too large. Note: Because this value is a system value, it will only be applied when the Communications Systems service is restarted.

Verify Client, require verification of client certificate (TLS ONLY).

Verify Server, require verification of server certificate (TLS ONLY).

Nat Settings Section

Local Net, network to consider local (used for NAT purposes). **External Media Address**, external IP address to use in RTP handling. **External Signal Address**, external address for SIP signaling.

7.1.2 SIP Settings

The SIP Settings is used to configure the default value to be used for SIP calls.

General

The screenshot shows the 'General' tab of the SIP Settings configuration page. The page has a top navigation bar with tabs: GENERAL, SECURITY, NETWORK, CODECS, OTHERS, and CUSTOM. The 'GENERAL' tab is active. The settings are organized into several sections:

- Language**: A dropdown menu set to 'Portuguese (Brazil) (pt_BR)'.
- Tone Zone**: A dropdown menu set to '(br) Brazil'.
- G726-32 Audio**: A toggle button set to 'No'.
- Notification Settings**:
 - Notify Ringing**: A toggle button set to 'Yes'.
 - Notify Hold**: A toggle button set to 'No'.
 - Notify CID**: A toggle button set to 'No'.
- Registration Settings**:
 - Max Expiry**: A text input field with the value '3600'.
 - Min Expiry**: A text input field with the value '60'.
 - Default Expiry**: A text input field with the value '120'.
 - MWI Expiry**: An empty text input field.
- Bind Address**: Two text input fields, the first containing '0.0.0.0' and the second containing '5062'.
- Allow Transfer**: A toggle button set to 'Yes'.
- Video Settings**:
 - Video Support**: A toggle button set to 'No'.
 - Max Call BitRate**: A text input field with the value '384'.
- Fax Settings**:
 - Fax Detect**: A toggle button set to 'No'.
 - T.38 Fax Pass-Through**: A toggle button set to 'No'.

At the bottom right of the form, there is a green 'Save' button with a floppy disk icon.

Language, default language setting for all users/peers. This may also be set for individual users/peers.

Tone Zone, default tone zone for all users/peers.

G726-32 Audio, if the peer negotiates G726-32 audio, use AAL2 packing order instead of RFC3551 packing order (this is required for Grandstream ATAs, among others).

Bind Address, IP address to which the device should bind. Note that 0.0.0.0 binds to all IP addresses.

Bind Port, port to which the device should bind. The standard SIP port is 5060.

Allow Transfer, disables all transfers (unless specifically enabled in peers or users). Default setting is enabled. Note that the dial options 't' and 'T' are not related as to whether SIP transfers are allowed or not.

Enable Websockets, set to "No" by default, so SIP Channels do not listen for a WebSocket. This is necessary when SIP and PJSIP WebSockets are used on the same system.

Notification Settings Section

Notify Ringing, controls whether busy subscribers get sent a RINGING message when another call is sent.

Notify Hold, notify subscribers that are in HOLD state.

Notify CID, controls whether caller ID information is sent together with dialog-info+xml notifications (supported by Snom phones).

Video Section

Video Support, this turns on or off support for SIP video. You need to turn this on to get any video support at all.

Max Call Bitrate, maximum bitrate for video calls (default 384 kb/s)

Registration Settings Section

Max Expiry, maximum allowed time, in seconds, for incoming registrations.

Min Expiry, minimum time, in seconds, for registrations.

Default Expiry, default length of incoming/outgoing registration.

MWI Expiry, expiry time for outgoing MWI subscriptions.

Security

GENERAL SECURITY NETWORK CODECS OTHERS CUSTOM

Allow Guest	No	Allow Msg Request	Yes
Failure Events	No	Disallow Dynamic Hosts	No
Always Reject	Yes	Allow External Domains	Yes
Auto-Domain	No		

SIP Domains

Domain
myasterisk.dom

Allow Guest, allow or reject guest calls. Do not activate this option unless you are sure what you are doing. By activating this option, you are allowing anyone to make a call to your PBX without having to register in it.

Failure Events, these generate manager peer-status events when peer cannot authenticate with Communications Systems. Peer-status will be rejected.

Always Reject, when rejecting an incoming INVITE or REGISTER call, for any reason, always reject with an identical response. This reduces the ability of an attacker to scan for valid SIP usernames.

Allow Msg Request, disable this option to reject all MESSAGE requests outside of a call. By default, this option is enabled, enabling MESSAGE requests to be passed to the dial-plan.

Disallow Dynamic Hosts, disallow all dynamic hosts from registering with any IP address used for statically defined hosts. This helps avoid the configuration error of allowing users to register with the same address as a SIP provider.

Auto Domain, this is an important setting with respect to SIP domains. When it is set to "no", Communications Systems will only recognize domains that were explicitly defined or will simply not support SIP domains at all (if there were no explicitly defined domains). If you set it to "yes" please be aware that

Communications Systems will create a domain based on the external IP address of your firewall as specified in the “externip” parameter. This might represent a compromise on your SIP security. If you don’t want a domain to be created based on “externip”, then set to “no” and explicitly add domains for your local (internal) IP address and for any other domains you require.

Allow External Domains, tells Communications Systems whether to allow SIP-to-SIP calls to non- local domains.

SIP Domains, when used, they provide enhanced security because registrations will only be accepted when they come from an IP phone (or other SIP client) that is using one of the recognized domains. When Communications Systems knows the identity of all its local SIP domains, this allows a higher level of security in the routing of SIP-to-SIP calls too.

Network

GENERALSECURITYNETWORKCODECSOTHERSCUSTOM

TCP EnableNo

TCP Bind Address0.0.0.05062

Enable TLSNo

TLS Bind Address0.0.0.05063

TLS Do Not VerifyNo

TLS CertificateMy Certificate

NAT

NATNo

External Address

External Host

External Refresh

Local Networks

IP Address

0.0.0.0

Network Mask

255.255.255.0

Add

Save

General Section

TCP Enable, use TCP.

TCP Bind Address, IP address and optional port to bind for this transport.

Enabled TLS, enable TLS.

TLS Bind Address, IP address and optional port to bind for this transport.

TLS Do Not Verify, if set to yes, don’t verify the server’s certificate when acting as a client.

TLS Certificate, the server’s certificate file.

NAT Section

NAT, NAT (Network Address Translation) is a technology most commonly used by firewalls and routers to allow multiple devices on a LAN with “private” IP addresses to share a single public IP address. A private IP address is an address, which can only be addressed from within the LAN, but not from the Internet outside the LAN.

Options:

- **No:** Do no special NAT handling other than RFC3581
- **Force:** Pretend there was a rport parameter even if there wasn't.
- **Comedia:** Send media to the port Communications Systems received it from regardless of where the SDP says to send it.
- **Auto Force:** Set the force_rport option if Communications Systems detects NAT.
- **Auto Comedia:** Set the comedia option if Communications

Systems detects NAT. **External Address**, specifies a static address, or address[:port] combination, to be used in SIP and SDP messages.

External Host, alternatively you can specify an external host, and Communications Systems will perform DNS queries periodically. Not recommended for production environments, Use "External Address" instead.

External Refresh, how often, in seconds, to refresh the "External Host," if used.

Local Networks Section

IP Address, valid IP address to be used.

Network Mask, network mask to use.

Codecs

GENERALSECURITYNETWORKCODECSOTHERSCUSTOM

Preferred Codec Only

No

Auto Framing

No

Available Codecs

Add all

slin

g726

gsm

ilbc

g723

g726aal2

adpcm

lpc10

speex

g722

h264

h263p

h263

h261

+

+

+

+

+

+

+

+

+

+

+

+

+

+

Selected Codecs

Remove all

ulaw

alaw

g729

×

×

×

Save

Preferred Codec Only, respond to a SIP invite with the single most preferred codec rather than advertising all codec capabilities. This limits the choice of codec by the remote side to exactly the codec that we prefer.

Auto Framing, this sets packetization based on the remote endpoint's (ptime) preferences.

Available Codecs, list of available codecs.

Selected Codecs, list of selected codecs.

Note: Click on + icons to move available codecs to list of selected codecs. Click on the x icon to remove codecs from the list of selected codecs. You can also select or remove all codecs by clicking on either the Add All or Remove All buttons.

Others

GENERAL
SECURITY
NETWORK
CODECS
OTHERS
CUSTOM

General

SRV Lookups

No

Qualify Frequency

SIP Debugging

SIP Debug

No

Record History

No

Dump History

No

SIP QoS

SIP TOS

cs3

Audio TOS

ef

Video TOS

af41

RTP Timers

RTP Timeout

30

RTP Hold Timeout

300

Jitter Buffer

Enabled

No

Force

No

Max Size

Resynchronized

Jitterbuffer implementation

Fixed

Target Extra

Save

General Section

SRV Lookups, enables or disables DNS SRV lookups on outbound calls.

Qualify Frequency, determines how often, in seconds, to check that the host is alive, as reported, in milliseconds, with sip show settings command.

RPT Timers Section

RPT Timeout, sets the time, in seconds, to terminate the call if there is no RTP or RTCP activity on the audio channel.

RPT Hold Timeout, sets the time, in seconds, to terminate the call if there is no RTP or RTCP activity on the audio channel.

SIP Debugging Section

SIP Debug, toggle SIP debugging, from the moment the channel loads this configuration.

Record History, toggles recording of SIP history.

Dump History, toggles dump SIP history at end of a SIP dialogue. SIP history is output to the DEBUG logging channel.

Jitter Buffer Section

Enabled, enables or disables the use of a jitter-buffer on the receiving side of a SIP channel.

Force, this forces the use of a jitter-buffer on the receiving side of a SIP channel.

Max Size, maximum length, in milliseconds, of the jitter-buffer.

Resynchronized, gap in the frame timestamps, in milliseconds, beyond which the jitter-buffer will be resynchronized.

Jitter-buffer Implementation, this is used on the receiving side of a SIP channel. There is a choice of two options:

Fixed: size always equals to jb-max-size

Adaptive: variable size, the new jb of IAX2.
Target Extra, this sets the time, in milliseconds, the new jitter buffer will pad its size. This option only affects the jb when 'jbimpl = adaptive' is set.

SIP QoS Section

SIP TOS, this sets the TOS for SIP packets.
Audio TOS, this sets the TOS for RTP audio packets.
Video TOS, this Sets the TOS for RTP video packets.

Custom

GENERALSECURITYNETWORKCODECSOTHERSCUSTOM

Custom Options

Parameter	Value

Add

Save

Custom Options Section

Parameter, the SIP parameter to be included in the [general] section.
Value, value of the SIP parameter to be used.

7.1.3 IAX2 Settings

The IAX Settings is used to configure the default value to be used for IAX calls.

General

GENERALREGISTRATIONCODECSSECURITY

Bind Address		ADSI	Yes
Bind Port	4569	SRV Lookup	No
Language	English (en)	Disable Checksums	Yes
Bandwidth	Low	IAX Compat	Yes

Save

Bind Address, IP address to which to bind. Note that 0.0.0.0 binds to all IP addresses.
Bind Port, port to which IAX2 should bind. The default port is 4570.
Language, this allows you to specify a global default language for all users who use this profile. This can be specified also on a per-user basis.
Bandwidth, here you can specify bandwidth (low, medium, or high) to control which codecs should be used.
ADSI, Analog Display Services Interface (ADSI) can be enabled if you have ADSI-compatible CPE equipment.

SRV Lookup, whether to perform SRV lookup on outbound calls.

Disable Checksums, disable use of UDP checksums. If no checksum is set, then checksum will not be calculated or checked on systems supporting this feature.

IAX Compact, set to yes if you plan to use layered switches or some other scenario which may cause a delay when performing a lookup in the dial-plan. This option causes Communications Systems to spawn a separate thread when it receives an IAX2 DPREQ (Dial-plan Request) instead of blocking while it waits for a response.

Registration

GENERALREGISTRATIONCODECSSECURITY

Minimum Expire	60	Auto Kill	yes
Maximum Expire	60	Trunk Frequency	20
IAX Thread Count	10	Authorization Debug	No
IAX Max Thread Count	100	Trunk Time Stamps	Yes

Save

Minimum Expire, minimum time, in seconds, that IAX2 peers can request registration.

Maximum Expire, maximum time, in seconds, that IAX2 peers can request registration.

IAX Thread Count, establishes the number of IAX helper threads to handle I/O.

IAX Max Thread Count, establishes the maximum number of IAX helper threads that can be used to handle I/O. The Communications Systems, establishes the number of extra dynamic threads that may be spawned to handle I/O.

Auto Kill, this is used to keep the system from stalling when a host is not available. In addition to 'yes' or 'no' you can also specify several milliseconds.

Trunk Frequency, sets how frequently, in milliseconds, trunk messages are sent. This means the trunk will send all the data queued to it in the past period. By increasing the time between sending trunk messages, the trunk's payload size will increase as well.

Authorization Debug, this enables authentication debugging, but will increase the amount of debug traffic.

Trunk Time Stamps, should we send timestamps for the individual sub-frames within trunk frames? There is a small bandwidth use for these (less than 1kbps/call), but they ensure that frame timestamps get sent end-to-end properly.

Codecs

GENERAL REGISTRATION CODECS SECURITY

Codec Priority Caller

Available Codecs

Add all	
adpcm	+
codec2	+
g722	+
g723	+
g726	+
g726aal2	+
gsm	+
h261	+
h263	+
h263p	+
h264	+
ilbc	+
lpc10	+
opus	+

Selected Codecs

Remove all	
ulaw	x
alaw	x
g729	x

Save

Codec Priority, this controls the codec negotiation of an inbound IAX2 call. There are several options:

- **Caller:** consider the preferred order of the caller before considering the preferred order of the host.
- **Host:** consider the preferred order of the host before considering the preferred order of the caller.
- **Disabled:** disable the consideration of codec preference altogether.
- **Reqonly:** Behaves in a similar manner as the disabled option. The call will only be accepted if the requested format is available.

Codecs Selection Section

Available Codecs, list of available codecs.

Selected Codecs, list of selected codecs.

Note: Click on + icons to move available codecs to list of selected codecs. Click on the x icon to remove codecs from the list of selected codecs. You can also select or remove all codecs by clicking on either the Add All or Remove All buttons.

Security

GENERAL REGISTRATION CODECS SECURITY

Call Token
Optional

Max Non-validated Call Nos

Max Call Numbers

Call Number Limits

IP Address	Mask	Limit	
0.0.0.0	255.255.255.0	100	

Add

Save

Call Token Optional, call token validation can be set as optional for a single IP address or an IP address range by enabling this option. This is a global option.

Max Call Numbers, this option limits the number of call numbers allowed for each individual remote IP address. Once an IP address reaches its call number limit, no more new connections are allowed until an existing connection is closed. This option can be used in a peer definition as well, but only takes effect for the IP of a dynamic peer after it completes registration.

Max Non-validated Call Nos, they parameter is used to set the combined number of call numbers that can be allocated for connections where call token validation has been disabled. Unlike the Max Call Numbers option, this limit is not separate for each individual IP address. Any connection resulting in a non-call token validated call number being allocated contributes to this limit.

Call Number Limits Section

IP Address, valid IP address to be used.

Mask, network mask to use.

Limit, this limits the number of call numbers allowed for this group of IP addresses. Once an IP address group reaches its call number limit, no more new connections are allowed until an existing connection is closed.

7.1.4 Device Profiles

Profiles are sets of characteristics that are generally repeated when creating extensions and / or devices. Instead of repeating in the forms and dial-plan these data we create profiles that group this data.

There are four profile type options: PJSIP SIP IAX2 Telephony (BRI, E1 PRI, E1 R2, FXO, FXS, T1 CAS, T1 PRI). These options will only appear if the DAHDI add-on is installed.

7.1.4.1 PJSIP Profile General

GENERALADVANCED

Profile Type

PJSIP IAX2

General

Name *

Description

Network

Transport

Auto

Quality Frequency

50

Quality Timeout

5

Default Expiration

3600

Maximum Expiration

7200

Minimum Expiration

600

Outbound Proxy

Force rport

Yes

ICE Support

No

Reinvite Contact

Yes

Remove Existing

Yes

Media

Media Encryption

No

Max Video Streams

5

Max Audio Streams

0

Direct Media Glare Mitigation

None

RTP Keepalive

15

RTP Timeout

50

DTLS

DTLS Certificate

None

DTLS Setup

Actpass

DTLS Verify

No

Use AVPF

No

RTP Symmetric

Yes

RTCP Mux

No

Asymmetric RTP Codec

Yes

Send Offerion Header

Yes

Send P-Asserted-Identity

Yes

Send Remote-Party-ID

Yes

Support Path

Yes

Send Connected Line

No

WebRTC

No

RTP Timeout Hold

300

Direct Media

Yes

Media Transport Received

Yes

Media Encryption Optomitic

Yes

Disable NAT Direct Media

Yes

DTLS Fingerprint Hash

SHA-256

DTLS Relay Interval

0

Save

Name, this is the user-defined name for the profile.

Description, a brief description to identify the profile.

Network

Transport, desired transport configuration.

Qualify Frequency, interval between attempts to rate the contact to reach it. If 0 never qualify. Time in seconds.

Qualify Timeout, If the contact does not respond to the OPTIONS request before the time out, the contact is marked as unavailable. If the value is 0 there is no timeout.

Force rport, force use of the return port.

ICE support, enable the ICE mechanism to help traverse NAT.

Rewrite Contact, this allows the contact header to be rewritten with source IP address port.

Remove Existing, this allows a registration to succeed by displacing any existing contacts that now exceed the "Max Contacts" count. Any removed contacts are the next to expire. The behavior is beneficial when "Rewrite Contact" is enabled and "Max Contacts" is greater than one. The removed contact is likely the old contact created by "Rewrite Contact" that the device is refreshing.

Use AVPF, determine if res_pjsip will use and enforce the use of AVPF for this endpoint.

RTP Symmetric, enforce that RTP must be symmetric.

RTCP Mux, with this option enabled, Communications Systems will try to negotiate the use of the "rtcp- mux" attribute on all media streams. This will result in RTP and RTCP being sent and received on the same port. This switches the demultiplexing logic to the application rather than the transport layer. This option is useful when interoperating with WebRTC endpoints as they enforce the use of this option.

Asymmetric RTP Codec, allow the send and receive RTP codec to differ.

Send Diversion Header, send the forward header, transmitting the forwarding information to the called user agent.

Send P-Asserted Identity, send the Header P-Asserted Identity.

Send Remote-Party-ID, send the Header Remote-Party-ID.

WebRTC, when set to "Yes", this also enables the following settings required for basic WebRTC support to work: `rtcp_mux`, `use_avpf`, `ice_support`, and `use_received_transport`.

Media

Media Encryption, this determines if res_pjsip will use and enforce the use of media encryption for this endpoint.

Direct Media, this determines if media can flow directly between endpoints.

Received Media Transport, this determines if res_pjsip will use the media transport received in the offer SDP in the corresponding response SDP.

Optimistic Media Encryption, this determines whether encryption should be used, if possible, but does not terminate the session if not achieved.

Disable NAT Direct Media, direct media session disable is updated when NAT obstructs media session.

DTLS

DTLS certificate, certificate to use with DTLS connections.

DTLS Setup, if we are willing to accept connections, connect with the other party, or both. Valid options are:

- **Active**, we want to connect with the other party.
- **Passive**, we only want to accept connections.
- **Actpass**, we will do both. This value will be used in outbound SDP when offered and for inbound SDP offers when remote party

sends actpass.

DTLS Verify, verify that the provided peer certificate is valid.

DTLS Fingerprint Hash, the hash to use for the fingerprint in SDP.

DTLS Rekey interval, interval in which to renegotiate the TLS session and reactivate the SRTP session. If this is not configured or the provided value is 0, reordering will be disabled.

Advanced

Here we can add any PJSIP parameter that is not included in the Attimo CloudPBX interface. And the options are as follows:

Parameter, the PJSIP parameter to be included in the PJSIP profile.

Value, value of the PJSIP parameter to be used.

7.1.4.2 SIP Profile

The screenshot shows the 'GENERAL' tab of the SIP Profile configuration. The settings are as follows:

- Language:** Portuguese (Brazil) (pt_BR)
- Tone Zone:** (br) Brazil
- G726-32 Audio:** No
- Notification Settings:**
 - Notify Ringing: Yes
 - Notify Hold: No
 - Notify CID: No
- Registration Settings:**
 - Max Expiry: 3600
 - Min Expiry: 60
 - Default Expiry: 120
 - MWI Expiry: (empty)
- Video Settings:**
 - Video Support: No
 - Max Call BitRate: 384
- Fax Settings:**
 - Fax Detect: No
 - T.38 Fax Pass-Through: No

At the bottom right, there is a green 'Save' button.

Name, user-defined name for the profile.

Description, short description to identify this profile.

Network

Host, the host parameter specifies the host name or IP address of a SIP peer or user. It is used to make outgoing calls and to find the pair when an incoming call is received. The host can take the following formats:

- Domain name / host name, eg.: sip.zxv.com
- IP address, eg. 234.23.42.103
- Dynamic, which means that phones must be registered.

Type, this determines their roles within Communications Systems. The type options are:

- **Peer:** they handle incoming and outgoing calls and correspond with IP / Port. When there are incoming calls from the peer, the IP address must match for the invitation to be accepted.
- **User:** They only handle incoming calls, which means they can call Communications Systems, but Communications Systems can't call

them. Callers must be authenticated by their authorization information (username and secret).

- **Friend:** will accept friend calls as it would for users, requiring only that the authorization match, rather than the IP address.

Qualify Frequency, how often the host should be verified to be in seconds and reported in milliseconds with the sip show setting.

Timeout for Qualifier, if this is set to yes (equivalent to 2000 ms), it will send an OPTIONS packet to the endpoint periodically (by default every minute). It is used to control the state of the endpoint. If the delays are longer than the qualify time, the endpoint will be disconnected and considered unreachable. It can be set to a value that is the threshold of msec. Setting to 'no' will not disable it. It may also be useful to keep NAT pinholes open.

Transport, this sets the default transports. The order determines the main default transport.

ICE, whether to enable ICE support. The default value is no. ICE (Interactive Connectivity Establishment) is a Network Address Translator (NAT) handover protocol for UDP-based multimedia sessions established with the offer / response model. This option is commonly enabled in WebRTC configurations.

RTP Encryption, enables RTP voice encryption.

AVPF, enable interoperability with media streams using the AVPF RTP profile.

Direct RTP, allows the direct sending of RTP between the call participants.

DTLS

Enable, enable, or disable DTLS-SRTP support.

DTLS Certificate, certified for use with DTLS connections.

DTLS Setup, if we are willing to accept connections, connect with the other party, or both. Valid options are:

- **Active**, we want to connect with the other party.
- **Passive**, we only want to accept connections.
- **Actpass**, we will do both. This value will be used in outbound SDP when offered and for inbound SDP offers when remote party sends actpass.

DTLS Verify, verify that the provided peer certificate is valid.

DTLS Fingerprint Hash, the hash to use for the fingerprint in SDP.

DTLS Rekey Interval, interval in which to renegotiate the TLS session and reactivate the SRTP session. If this is not configured or the provided value is 0, reordering will be disabled.

Caller ID (CID)

Send ID to Remote Location, add remote party ID header to SIP INVITATIONS.

Trust Remote Location ID, with this we assume the remote party ID header is correct.

Advanced

Here we can add any SIP parameter that is not included in the Attimo CloudPBX interface. And the options are as follows:

Parameter, the SIP parameter to be included in the SIP profile.

Value, value of the SIP parameter to be used.

7.1.4.3 IAX2 Profile

GENERAL

REGISTRATION

CODECS

SECURITY

Bind Address		ADSI	Yes
Bind Port	4569	SRV Lookup	No
Language	English (en)	Disable Checksums	Yes
Bandwidth	Low	IAX Compat	Yes

Save

Name, user-defined name for the profile.

Description, short description to identify this profile.

Network

Host, hostname, or device address.

Type, this defines the type of device.

- **User**, this option is a device that makes calls, requires authentication.
- **Peer**, this option is a trunk device, accompanied by the Host.
- **Friend**, this option is a combination of "User" and "Peer".

Test call, this uses require call token for authentication.

Qualifier Frequency, this defines the interval of qualifying in seconds. A value of zero will disable this feature.

Qualifier Timeout, this defines the maximum response time in milliseconds before a device is considered unreachable. A value of zero will disable this feature.

Transfer, allow transfers from this device.

Advanced

Here we can add any IAX2 parameter that is not included in the Attimo CloudPBX interface.

And the options are as follows:

Parameter, the IAX parameter to be included in the profile.

Value, value of the IAX parameter to be used.

7.1.5 Dial Profiles

All dialing options are grouped in a Dial Profile.

General

The screenshot shows the 'GENERAL' tab of a Dial Profile configuration page. It features several input fields and dropdown menus. On the left, there is a 'Name' field with an asterisk, followed by 'Allow Transfer By' and 'Allow Park By', both set to 'Recipient'. Below these is 'Ringback Tone' set to 'None (Ringback)'. On the right, 'Call Screening' is set to 'Disabled', 'Custom Options' is an empty text box, and 'Ringing Tone' has a blue 'Yes' button. A yellow warning box at the bottom states: 'Warning! Enabling transfer by caller may allow an outside caller to exploit the PBX by transferring their calls to another external destination when a too permissive Class of Service is defined for the trunk.' A green 'Save' button is at the bottom right.

Name *		Call Screening	Disabled
Allow Transfer By	Recipient	Custom Options	
Allow Park By	Recipient	Ringing Tone	Yes
Ringback Tone	None (Ringback)		

⚠ Warning! Enabling transfer by caller may allow an outside caller to exploit the PBX by transferring their calls to another external destination when a too permissive Class of Service is defined for the trunk.

Save

Name, name to identify this profile.

Allow Transfer By, allow the called/calling party to transfer the calling party by sending the DTMF sequence defined in feature codes.

Allow Park By, allow the called/calling party to enable parking of the call by sending the DTMF sequence defined for "Park Call" in feature codes.

MoH Class, provide hold music to the calling party until a requested channel answers. **Call Screening**, before ringing your extension, the caller is asked to supply an introduction. The application asks them: "After the tone, say your name". They are allowed 4 seconds of introduction.

Custom Options, this allows you to define custom dial parameters that have not been included. Examples of some common dial options:

- D (called:calling) - send the specified digits after the called party has answered, but before the call gets bridged. The 'called' digits are sent to the called party, and the 'calling' digits are sent to the calling party. Both arguments can be used alone.
- H - allow the called party to hang up by using the In-Call Communications Systems Disconnect code (default value is **)
- i - any forwarding requests that may be received on this dial attempt will be ignored.
- I - any connected line update requests or any redirecting party update requests that may be received on this dial attempt will be ignored.
- r - generate ringing to the calling party, even if the called party is not actually ringing. Pass no audio to the calling party until the called channel has answered.
- S(x) - hang up the call x seconds after the called party has answered the call.
- t - allow the called party to transfer the calling party by using the In- Call Communications Systems Blind Transfer code (default value is ##)
- T - allow the calling party to transfer the called party by using the

In- Call Communications Systems Blind Transfer code (default value is ##)

- w - allow the called party to enable recording of the call by using the In-Call Communications Systems Toggle Call Recording code (default value is *1)
- W - allow the calling party to enable recording of the call by using the In-Call Communications Systems Toggle Call Recording code (default value is *1)

Ringing Tone, this indicates a ringing tone to the calling party, even if the called party is not yet ringing. Do not pass audio to the calling party until the called channel is answered.

Warning! Enabling transfer by caller may allow an outside caller to exploit the PBX by transferring their calls to another external destination when a too permissive Class of Service is defined for the trunk.

7.2 Voicemail Settings

7.2.1 Voicemail Settings

In this tab you will find the information about Voicemail Settings.

General

GENERAL

EMAIL SETTINGS

Max Message Length	180	MP3 Attachments	No
Min Message Length		Move Heard Msg	Yes
Greetings Length	60	Force Name	No
Max Silence	10	Force Greetings	No
Max Login Attempts	3	Use Directory	Yes
Backup Deleted	100	Review Msg	Yes
Max Messages	100	Search Contexts	No
Locale	English (United States)	Email date format	%A, %B %d %Y, %R

 Save

Max Message Length, maximum length of a voicemail message in seconds. Leave empty for No Limit.

Min Message Length, minimum length of a voicemail message in seconds for the message to be kept. Leave empty for No Minimum.

Greetings Length, maximum length of greetings in seconds.

Max Silence, how many seconds of silence before we end the recording.

Max Login Attempts, max number of failed login attempts.

Backup Deleted, maximum number of messages allowed in the Deleted folder.

Max Messages, maximum number of messages per folder. If set to 0, a mailbox will be greetings-only.

Operator Number, Extension to dial on pressing 0 while listening a voicemail.

Move Heard Msg, move heard messages to the Old folder automatically.

Force Name, forces a new user to record their name. A new user is determined by the password being the same as the mailbox number.

Force Greetings, this is the same as "Force Name", except for recording.

Use Directory, permit finding entries for forward/compose from the directory.

Operator, this will allow the sender to hit 0 before/after/during leaving a voicemail to reach an operator. This option **REQUIRES** an o extension in the same context (or in exit context, if set), as that is where the 0 key will send you.

Review Msg, allow sender to review/rerecord their message before saving it

Search Context, current default behavior is to search only the default context if one is not specified. The older behavior was to search all contexts.

Operator Destination, the extension to call when pressing 0 while listening to the voicemail box.

Time Definition*, users may be in different time zones, or may have different message announcements for their introductory message when they enter the voicemail system. Set the message and the time zone each user hears here. Set the user into one of these zones with the tz= attribute in the options field of the mailbox. Of course, language substitution still applies here so you may have

several directory trees that have alternate language choices. Look in /usr/share/zoneinfo/ for names of time zones.

Supported values:

- 'filename', filename of a sound file (single ticks around the filename required).
- \${VAR}, variable substitution.
- A or a, Day of week (Saturday, Sunday, ...).
- B or b or h, Month name (January, February, ...).
- d or e, numeric day of month (first, second, ..., thirty-first).
- Y, Year.
- I or I, Hour, 12-hour clock.
- H, Hour, 24-hour clock (single digit hours preceded by "oh").
- K, Hour, 24-hour clock (single digit hours NOT preceded by "oh").
- M, Minute, with 00 pronounced as "o'clock".
- N, Minute, with 00 pronounced as "hundred" (US military time).
- P or p, AM or PM.
- Q, "today", "yesterday" or ABdY (*note: not standard strftime value).
- Q, "" (for today), "yesterday", weekday, or ABdY (*note: not standard strftime value).
- R, 24-hour time, including minute.

Examples:

```
eastern=America/New_York|'vm-received' Q 'digits/at' IMp
central=America/Chicago|'vm-received' Q 'digits/at' IMp
central24=America/Chicago|'vm-received' q 'digits/at' H N 'hours'
military=Zulu|'vm-received' q 'digits/at' H N 'hours' 'phonetic/z_p'
european=Europe/Copenhagen|'vm-received' a d b 'digits/at' HM
```

7.3 PBX Settings

7.3.1 System General

In this tab you will find the information about PBX Settings

General

GENERALSYSTEM PROMPTSSYSTEM DIRECTORIES

Extension Settings

Default Language

Portuguese (Brazil) (pt_BR)

Devices Prefix

Public Domain

Auto-Generated Passwords Length

25

Enable Voicemail

Yes

Enable Portal

No

Create Hints

No

Email Credentials

Yes

Send Diversion Headers

No

Dial-Plan Settings

Default Ring Time

30

Attended Transfer Timeout

15

Transfer Digit Timeout

3

Features Digit Timeout

1000

Recording Script

/var/lib/my_script

Drop Attended Transfer Callbacks

No

Play Call Waiting Tone

No

Route Calls to Tenant DID's Internally

No

GUI Settings

Show Add-ons Menu

Yes

Queues CallBack

Search Frequency

300

Save

Extension Settings

Default Language, language to select by default when a new extension is being created.

Device Prefix, prefix to append by default to devices name.

Enable Voicemail, if enabled, the voicemail will be enabled automatically each time you create an extension.

Enable Portal, if enabled, the portal will be enabled automatically each time you create an extension.

Create Hints, if enabled, the hints will be enabled automatically each time you create an extension.

Dial-Plan Settings

Default Ring Timer, time extensions by default ringing.

Transfer Digit Timeout, number of seconds to wait between digits when transferring a call (default is 3 seconds).

Features Digit Timeout, max time (ms) between digits for feature activation (default is 1000 ms).

Recording Format, file format for calls recording.

Recording Script, Script to be executed when the recording is over. The script parameters can be defined as space separated sequence of strings like `^{name}`, where name can be any channel or MixMonitor variable. For example, `^{UNIQUEID}` - channel ID, `^{MIXMONITOR_FILENAME}` - recording file name.

GUI Settings

Show Login Panel, if set to yes, it shows a panel with the sonata add-ons when these are installed.

Queues Callback

Search Frequency, this value defines the frequency in which the Queues Callback service will search for calls.

System Prompts

It allows to customize certain voice guides of the system, now only a couple of options are possible.

GENERAL

SYSTEM PROMPTS

SYSTEM DIRECTORIES

DND

Default

Blacklisted

Default

Save

DND, allows you to define a custom prompt that will be reproduced when an extension has active the DND.

Blacklisted, allows you to define a custom prompt that will be reproduced to callers who are blacklisted.

System Directories Tab

GENERAL

SYSTEM PROMPTS

SYSTEM DIRECTORIES

Description	Path
AGI Directory	/var/lib/asterisk/agi-bin
Directory	/etc/asterisk
Log Directory	/var/log/asterisk
Modules Directory	/usr/lib/asterisk/modules
Sound Directory	/var/lib/asterisk/sounds
Spool Directory	/var/spool/asterisk
Libraries Directory	/var/lib/asterisk

Save

Directories

AGI Directory, AGI directory. Directory, home directory.

Modules Directory, modules directory. Libraries Directory, libraries directory. Log Directory, log directory.

Sound Directory, sound directory.

Spool Directory, spool (recording) directory.

7.3.2 AMI

In this tab you will find the information about Communications Systems Manager Users

General

GENERAL

AMI User *

AMI Secret *

ESyJUYfNuK37eB

Description *

Read Permissions

Deny *

0.0.0.0/0.0.0.0

Permit *

127.0.0.1/255.255.255.0

Write Permissions

Save

AMI User*, username for connect to AMI.

AMI Secret, secret for login to AMI.

Description, short description.

Read Permissions, AMI read permissions.

Write Permissions, AMI write permissions.

Deny, if you want to deny many hosts or networks, use & char as separator. Example:
192.168.1.0/255.255.255.0&10.0.0.0/255.0.0.0

Permit, if you want to permit many hosts or networks, use & char as separator.

7.3.3 Log File

In this tab you will find the information about create new log file.

General

GENERAL

Date Format

%F %T

Log Rotation

Sequential

Append Hostname

No

Log Files

Filename	Debug	DTMF	Error	Fax	Notice	Verbose	Warning	Security	
fail2ban	Off	Off	Off	Off	On	Off	Off	On	
console	On	Off	On	Off	On	3	On	Off	
full	On	Off	On	Off	On	3	On	Off	

Add

Save

Date Format, here you can customize the display of debug message time stamps. See strftime(3) Linux manual for format specifiers. Note that there is also a fractional second parameter which may be used in this field. Use %1q for tenths, %2q for hundredths, etc. Leave blank for default: ISO 8601 date format yyyy-mm-dd HH:MM:SS (%F %T).

Log Rotation, how to write down the logs.

- **Sequential:** Rename archived logs in order, such that the newest has the highest sequence number.
- **Rotate:** Rotate all the old files, such that the oldest has the highest sequence number (expected behavior for Unix administrators).
- **Timestamp:** Rename the log files using a timestamp instead of a sequence number when "logger rotate" is executed.

Append Hostname, appends the hostname to the name of the log files.

Log Files

File Name, name of log file.

Debug, debugging is only useful if you are troubleshooting a problem with the Communications Systems code itself. You would not use debug to troubleshoot your dial-plan, but you would use it if the Communications Systems developers asked you to provide logs for a problem you were reporting. Do not use debug in production, as the amount of detail stored can fill up a hard drive in a matter of days.

DTMF, logging DTMF can be helpful if you are getting complaints that calls are not routing from the auto attendant correctly.

Error, errors represent significant problems in the system that must be addressed immediately.

Notice, you will see a lot of these during a reload, but they will also happen during normal call flow. A notice is simply any event that Communications Systems wishes to inform you of.

Verbose, this is one of the most useful of the logging types, but it is also one of the riskier to leave unattended, due to the possibility of the output filling your hard drive. **Warning**, a warning represents a problem that could be severe enough to affect a call (including disconnecting a call because call flow cannot continue). Warnings need to be addressed.

Security, output security messages.

7.3.4 RTP Settings

In this tab you will find the information about RTP Settings

General

GENERAL

RTP Start	10000	Stun Server	
RTP End	20000	Turn Server	
Strict RTP	No	Turn Server Name	
RTP Checksums	No	Turn Server Password	
ICE Support	Yes		

ICE Host Settings

Local Address	Advertised Address	Include Local Address	
10.253.10.27	119.8.85.137	Yes	Add

RTP Start, indicates where the port start.

RTP End, indicates the end of the port.

Strict RTP, enable strict RTP protection. This will drop RTP packets that do not come from the source of the RTP stream, whether to enable or disable UDP checksums on RTP traffic.

RTP Checksums, whether to enable or disable UDP checksums on RTP traffic.

ICE Support, whether to enable or disable ICE support. This option is disabled by default.

Stun Server, hostname or address for the STUN server used when determining the external IP address and port an RTP session can be reached at. The port number is optional.

Turn Server, hostname or address for the TURN server to be used as a relay. The port number is optional.

Turn Server Name, username used to authenticate with TURN relay server.

Turn Server Password, password used to authenticate with TURN relay server.

7.3.5 CEL Settings

The Channel Event Logs for Communications Systems provides a mechanism to track various events related to a call. CEL es quite granular and a fine grain at that. It has been designed having billing information in mind. It admits various backend for storage and is a great alternative for the Call Detail Recordings for administrators that require event registries that are extremely detailed. The immense amount of detail will allow for the construction of precise data for billing and callflow.

General

GENERAL

Enable

No

Events

APP_START,CHAN_START...

APPS

Dial,Park,Queue,ConfBrid...

Date Format

%F %T

Save

Enable, enable or disable the Channel Event Logs for Communications Systems.
APPS, this allows you to specify the list of applications from which you want to receive the CEL events from.
Events, this allows you to specify a list of events that you wish to generate when they occur.
Date Format, this allows you to specify the date format used when generating CEL Events.

7.3.6 Mini HTTP Server

The core of Communications Systems provides a basic HTTP/HTTPS server. Certain Communications Systems modules may make use of the HTTP service, such as the Communications Systems Manager Interface over HTTP, the Communications Systems Restful Interface or WebSocket transports for modules that support that, like chan_sip or chan_pjsip.
In this tab you will find the information about Mini HTTP Server.

General

GENERAL

HTTP Bind Address

0.0.0.0

8088

TLS Bind Address

0.0.0.0

8089

Certificate

-- None --

Sessions Limit

1000

Enable HTTP

No

TLS Enable

No

Save

HTTP Bind Address, address and optional port to bind to, both for HTTP and HTTPS.
TLS Bind Address, address and port to bind to this transport.

Certificate, certificate for TLS connections.

Enable HTTP, whether HTTP/HTTPS interface is enabled or not.

TLS Enable, HTTPS support. In addition to enabled, you need to explicitly enable TLS, define the port to use, and have a certificate somewhere.

7.4 Voice Prompts

7.4.1 PBX Sounds

This module allows end users to install additional Communications Systems sounds according to their needs.

GENERAL

Sound Package	Installed Version	Available Version	Actions
German		4.0.0-3	
English	4.0.0-4		
English (Australia)		4.0.0-3	
English (United Kingdom)		4.0.0-3	
Spanish (Mexico)	4.0.0-3		
French		4.0.0-3	
Italian		4.0.0-3	
Portuguese (Brazil)	4.0.0-1		 

 Check Online

7.4.2 Music on Hold

In this tab you will find the information about Music on Hold. As an extended feature, you will be able to use an application to playback the MoH. This can be used to stream music from a streaming server.

General

GENERAL

Name *

Mode

Files

Sort

Linear

Sound file



Default

No



 Save

Name, short description for identify this MoH Category.

Mode, this defines the mode to play or retrieve music on hold.

- **Options:**

- **Files:** plays files from a directory in any media format supported by Communications Systems.
- **Custom:** run a custom application. For example, an online radio.

Sort, sort the files to listen to.

Sound file, you can use this to upload a wav or mp3 file.

Default, if checked the sounds from this music group will be used for the default

music class on hold. Only one music group can be marked as default.

Application, application that will be used to reproduce a broadcast. If the application is not provided, the mpg123 application will be used in the following format: mpg123 - q -r 8000 -f 8192 --mono -s.

Transmission URL, URL that will be transmitted by the defined application.

Format, the format option specifies the audio format that the application will provide to Communications Systems.

7.4.3 Recording Managements

In this tab you will find the information about Record Management.

General

GENERAL

Name *

Recording Code

Sound File *

Authorization PIN

Recording List

Show 10 entries

Search:

Name	Recording	Recording Code	Duration	Actions
No data available in table				

Showing 0 to 0 of 0 entries

Previous

Next

Upload Recording

Refresh

Name, short description for identify this recording.

Sound File, here you can upload a wav or mp3 file.

Recording List

Recording, name of the sound file. **Name**, description of the sound file. **Duration**, duration of the recording. Action

-  , edit description.
-  , listen recording.
-  , record by phone.
-  , delete recording.

8. Admin

8.1 Admin

8.1.1 Users

In this tab you will find the information about management users.

General

GENERAL

SETTINGS

Username *

E-mail

Password *

Profile *

Startup Dialog *

Full Name

Department

Tenants *

Enable 2FA

Super Administrator

Dashboard

Attimo

No



Select Image

Save

Username *, user as you will be login (Nick name).

E-Mail, this is the email address for the user. **Password ***, your secure password for login **Profile ***, profile for this user.

Startup Menu *, set the module for startup.

Full Name, full real name user.

Department, user department (Example: Admin)

Tenants *, this option allows you to select which tenants a user will have access to.

Select Image, image with which the user is associated, it can be a photo of the user.

Note: The main user, "admin", can only have its privileges modified given they are always total. If you want to create a user with administrator privileges, select the "Super Administrator" profile.

Settings

GENERAL

SETTINGS

GUI Theme

Language

Timezone

Attimo

English (en_US)

System

Visible By Others

Multitab

Dark Mode

Yes

Yes

No

Save

GUI Theme, appearance of the interface.

Language, interface language.

Time Zone, local Time Zone for this user.

Multi-tab, you can enable this if you want multiple tabs in the interface with modules you previously visited.

Dark Mode, when this option is enabled Attimo CloudPBX's interface will turn into a dark

theme. Great for working in the dark and saving a mobile device's battery life.

8.1.2 Users Profiles

In this tab you will find the information about management user's profiles.

General

The screenshot shows the 'General' tab of the 'Users Profiles' management interface. At the top, there are two tabs: 'GENERAL' (selected) and 'PERMISSIONS'. A hamburger menu icon is in the top right corner. Below the tabs, there is a 'Profile Name' label with a red asterisk and an empty text input field. Further down, there is a 'Module Access' section. It contains two buttons: 'Expand All' (green) and 'Select All' (blue). Below these buttons is a list of modules with radio buttons: PBX, Reports, Settings, Admin, and Portal. At the bottom right of the form, there is a green 'Save' button with a floppy disk icon.

Profile Name, name for this profile.

Modules Access

Grant Privileges, this option allows access to all modules.

Permissions

By its due nature, some permissions are not linked to the main menu. On this form you can configure these types of permissions.

The screenshot shows the 'Permissions' tab of the 'Users Profiles' management interface. At the top, there are two tabs: 'GENERAL' and 'PERMISSIONS' (selected). A hamburger menu icon is in the top right corner. Below the tabs, there are two rows of settings. The first row has 'Update Profile' with a blue 'Yes' button and 'Change Language' with a blue 'Yes' button. The second row has 'Get Updates' with a grey 'No' button. At the bottom right of the form, there is a green 'Save' button with a floppy disk icon.

Update Profile, this determines if a user with this profile can update their profile information, with the only limitation being to change the User Profile.

Get Updates, if a user with this profile has this option, they will be able to update Attimo CloudPBX from the GUI.

Change Language, this determines if a user with this profile can change the language from the GUI.

8.1.3 Application Keys

Here, we can create the API Key to grant permissions to third-party applications to use the Attimo CloudPBX API.

General

GENERAL

Description

Tenant

Attimo

Enabled

Yes

Save

8.1.4 Tenants

The term "software multitenancy" refers to a software architecture in which a single instance of software runs on a server and serves multiple tenants. A tenant is a group of users who share a common access with specific privileges to the software instance. With a multitenant architecture, a software application is designed to provide every tenant a dedicated share of the instance - including its data, configuration, user management, tenant individual functionality and non-functional properties.

As of version 2.2 of Attimo CloudPBX, the Multitenant module is added with which, for free, you can have the main Tenant plus an additional one to test all the functions.

General

GENERALCALLS ROUTINGSETTINGS

Name *

Description *

Prefix

Generate Automatically

Enabled

Yes

Tenant Administrator

Assign to Existing User

No

Admin Email *

Admin Password *

Full Name

Profile *

Tenant Administrator

Startup Dialog *

Dashboard

Send Welcome Email

Yes

Privileges

Extensions

Unlimited

Trunks

Unlimited

Queues

Unlimited

IVRs

Unlimited

Conferences

Unlimited

Parking Lots

Unlimited

Allow Recordings

No

Save

Name, a unique name for this tenant. This name will be used to create folders, linking CDR info, etc.

Description, this is a short Description to identify this tenant.

Prefix, this allows you to define a prefix to be used for extensions devices and others. If left blank an automatic prefix will be used.

Enabled, it allows you to enable/disable a tenant. If the tenant is disabled, the users who belongs to it will not be able to login to it nor perform any action.

Tenant Administrator

Assign to Existing User, if checked, instead of creating a new user for the tenant, you may assign it an existing one.

Admin Email, the email address of the user who will manage this Tenant.

Admin Password, password to authenticate the default admin user of this tenant.

Full Name, administrator's full name, if not defined, the tenant description will be used instead.

Profile, role profile for the administrator of this tenant. **Be careful not to assign a too permissive role, which may affect other tenants.**

Startup Dialog, which dialog to be displayed when logging into the system.

Limitations

Extension, it allows you to define the maximum number of extensions for this tenant.

Trunks, it allows you to define the maximum number of trunks for this tenant.

Queues, it allows you to define the maximum number of queues for this tenant.

IVRs, it allows you to define the maximum number of IVRs for this tenant.

Conferences, it allows you to define the maximum number of conferences for this tenant.

Parking Lots, it allows you to define the maximum number of parking lots for this tenant.

Softphone Devices, it allows you to define how many Sonata Communicator/Attimo CloudPBX Communicator Devices could be activated on this tenant.

Allow Recordings, it allows you to define if this tenant will be able to record or not calls.

Recordings Maintenance Settings

Clear Oldest Recordings, this allows you to define the maximum number of days that recordings should be retained. The recordings with more age than the days defined here will be deleted.

Schedule, it allows you to define the schedule in which the maintenance of the PBX will be executed (Conversion of Recordings, cleaning of Recordings and CDR, etc). If no schedule is selected, all the maintenance options will be disabled.

Convert Recordings, Enabled/Disable call recordings conversion to MP3.

Call Routing

It is possible to share the route selections items as outbound profiles for other tenants, this way you will not need to create tenant trunks for using main tenant as a gateway, and not need to re-define outbound routes per tenant.

GENERAL CALLS ROUTING SETTINGS

Outbound Profiles

Restricted CID

Outbound Calls Limit

Inbound Calls Limit

Default External CID

Emergency Trunks

Allowed Tenant Trunks

Allowed Outbound Routes

Inbound DIDs

DID Pattern	Description
1NXXNXXXXXX	Optional Description

Add

Save

Allow Tenant Trunks, it allows you to define which tenants could be used as tenant trunks.

Allow Outbound Routes, routes that will be used when this tenant make calls through a tenant trunk pointing to the main tenant. Calls made to any other tenant than the main tenant will be sent through Inbound Routes definitions.

Outbound Profiles, this allows you to define what route selection items can be used as an outbound profile on this tenant. From version 3 and onward, when you assign an Outbound Route, it is automatically applied to the All Permissions Class of Service for the Tenant. To create an Outbound Route, you must create a Route Selection under PBX > Class of Service > Route Selection.

Emergency Trunks, this option allows you to share emergency trunks with the tenant for the Emergency Numbers module.

Inbound DIDs, list of numbers/patterns belonging to this tenant. Calls that match with these numbers will be forward automatically to these tenant inbound routes. The configuration of these numbers takes precedence over the inbound routes' definitions on the main tenant. From version 3 and onward, when you assign a DID for the Inbound DID's section on the tenant, this will automatically create an Inbound Route on the tenant, where the Destination will be "Verify DID" which allows you to hear the DID number back when dialed.

Settings

GENERAL CALLS ROUTING **SETTINGS**

Time Zone System Disable Trunks Prefix No

External Add-ons

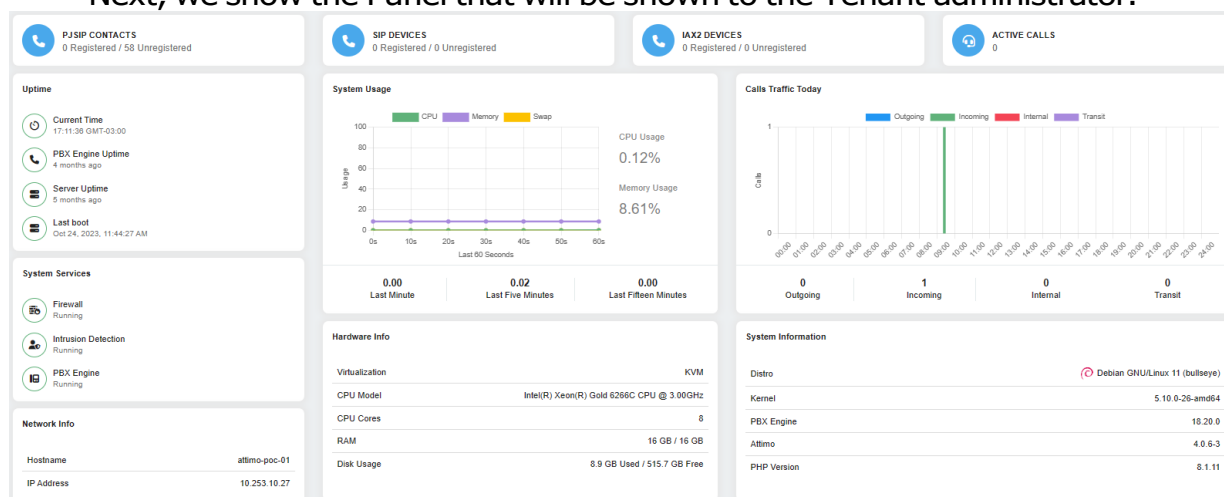
Save

It is possible from version 3 of Attimo CloudPBX and onward to further customize the Tenant experience. Here you can do the following.

Time Zone, select the time zone that this tenant will operate in. This will affect every time-based configuration for the tenant.

External Add-ons, here you can select which external add-ons you want to give the tenant access to. External add-ons would be the Sonata Suite of Applications and VitXi.

Next, we show the Panel that will be shown to the Tenant administrator.



8.2 System Settings

8.2.1 System Miscellaneous

In this tab you will find the information about System Miscellaneous Settings.

General

GENERAL

TIME

NOTIFICATIONS

STORAGE

⏻ Power Off

🔄 Restart

Save

⏻ Power Off

, server power off (be careful).

🔄 Restart

, restart Communications Systems.

Time

GENERAL

TIME

NOTIFICATIONS

STORAGE

Time Zone

(GMT -3:00) America/Sao_P...

Date

2024-03-18

Time

11:10:23

NTP

Yes

Server 1

0.debian.pool.ntp.org

Server 2

1.debian.pool.ntp.org

Server 3

2.debian.pool.ntp.org

Server 4

3.debian.pool.ntp.org

Clock Status

🔄 Refresh

Save

Time Zone, this sets the server time zone.
Date, here you can set date of the system.
Time, here you can set time of the system.
NTP, Enable/Disable NTP.
Server1, NTP server.
Server2, NTP server.
Server3, NTP server.
Server4, NTP server.

Notifications

GENERAL

TIME

NOTIFICATIONS

STORAGE

From Address

Intrusion Email

Storage Notifications

Save

From Address, the address entered here will be set as the "From:" address.

Storage Notifications, this will send notifications if your hard drive become low on space, or the raid has been broken.

Intrusion Detection, this will send notifications if a remote Ip has been banned from your system.

Abnormal call Volume, this will send notifications to notify you of abnormal call volumes.

Storage

GENERAL

TIME

NOTIFICATIONS

STORAGE



/ => 8 GB Used Of
492 GB (1.68%)

70

Enabled Disabled

Save

8.2.2 Email Settings

Server allows you the option to send outbound email messages either by using the built-in mail server (such as Postfix) that is active on the PBX server, or by using a network-accessible relay server that is hosted on another machine. Click on Use Built- in Mail Server to use the built-in mail server that is active on PBX or Use External Mail Server to use a network-accessible relay server.

In this tab you will find the information about email settings.

General

GENERAL

Server

Use Built-in Mail Server

Use External Mail Server

From Address

Hostname

Origin

Domain

Test Saved Email Settings

Enter an e-mail address to test the settings

Email Log

Refresh

Save

Server, you can relay outbound email messages either by using a built-in mail server (such as Postfix) that is active on the Attimo CloudPBX server, or by using a network-accessible relay server that is hosted on another machine.

From Address, the address entered here will be set as the "From:" address.

Provider, provider can be Gmail, or any other provider. The only additional information required for Gmail is Username and Password.

SMTP Server, the address that your provider has given you to enable you to send outgoing emails.

Port, SMTP server IP port number. By default, port 25 is used.

Origin, this specifies the origin domain for all mail posted by the PBX. By default, origin is configured to use your server hostname (e.g., pbx.mycompany.com)

TLS, Transport Layer Security (TLS) is a protocol that ensures privacy between communicating applications and their users on the Internet. When a server and client communicate, TLS ensures that no third party may eavesdrop or tamper with any message. TLS is the successor to the Secure Sockets Layer (SSL).

Authentication, here you can select the Use Authentication button if your email provider requires authentication (Username and Password) to send outgoing email. **Username**, username that your email service provider has given you to allow you to access your email account. Typically, this would be something like user@my-mail-server.com.

Password, the password that you use to log into your email account.

Testing Email Settings, testing the email settings.

Email Log, display email event log.

8.2.3 Email Templates

General

GENERAL	
Notification Type	Actions
New Voicemail Email	 
Extension Welcome Email	 
Emergency Notification Email	 
Tenant Welcome Email	 
Missed Call	 

From Version 3 and onward, you can send various email notifications, and on this module, you can customize the email templates. You can customize the following types of email notifications.

New Voicemail Email, here you can customize the email notification sent when using the voicemail to email feature.

Extension Welcome Email, here you can customize the email notification sent to new extensions. This is the same email sent when you select “Send Welcome Email” on the Extensions Module.

Emergency Notification Email, this is the email to be sent to the Emergency Contacts under the Emergency Numbers Module.

Tenant Welcome Email, this is an email sent to the tenant administrator whenever a new tenant is created and assigned to them.

Phonebooks QR Email, this is the email sent when you press the “Email QR code”.

option on the Phonebooks module. You can then perform a couple of actions. These would be to edit the email templates and to send a test email, so you can verify the email’s design.

8.2.4 Certificates

In this tab you will find the information about Certificates.

General

GENERAL 

Type

Self Signed 

Description *

Hostname *

Type, certificate type, Self Signed, Let’s Encrypt or Custom

Description, short description to identify this certificate.

Hostname, name of hostname

Owner Email, Owners Email (Let’s Encrypt only).

Country, Country (Let’s Encrypt only).

State, State (Let's Encrypt only).

Certificate, this should be the content of certificate file (e.g.: certificate.crt) (Custom only).

Key, this should be the content of key file (e.g.: private.key) (Custom only).

Chain, this should be the content intermediate certificate file (e.g.: ca_bundle.crt) (Custom only).

8.2.5 HTTP Server

In this tab you will find the information about HTTP Server settings.

General

GENERAL

HTTP Port	80	HTTPS Enable	<input checked="" type="button" value="Yes"/>
HTTPS Port	443	Force HTTPS	No
Certificate	My Certificate ▼		

HTTP Port, it defines from which port will be accessible the GUI through HTTP protocol.

HTTPS Port, it defines from which port will be accessible the GUI through HTTPS protocol.

Certificate, it defines the certificate to use when the GUI be accessed through HTTPS. By default, the pre-build certificate is used, if you want to use another go to Certificates module to generate one.

HTTPS Enable, if checked the GUI will be accessible through HTTPS. Enable by default. **Force HTTPS**, if checked all the traffic to HTTP protocol it will be redirected to HTTPS protocol.

8.3 Firewall

8.3.1 Settings

Attimo CloudPBX comes with a pre-configured firewall built in. You can choose to enable or disable the Firewall from here.

Intrusion Detection configures the Fail2ban application, which detects unauthorized attempts to access the system. After detecting the possible intruder, the IP Address from the intruder will be banned to prevent any further access attempts for a period.

A possible intrusion is defined by a specified quantity of failed attempts within a specific time frame defined in seconds. An alert can be sent to a defined email address to alert about the possible intrusion once it is detected.

You can create a whitelist of the IP Addresses that must be ignored by this application. Usually, you must include the PBX on the Whitelist, by adding the localhost 127.0.0.1 to the Whitelist Section.

In this module you will find information about the status of the firewall.

General

GENERAL

Enable Firewall

Yes

Enable
Intrusion
Detection

Yes

Intrusion Detection Settings

Failed Attempts
Allowed

3

Ban Time

86400

Find Time

600

Notifications E-
mail

Enable Firewall, enable or disable the Firewall.

Enable Intrusion Detection, enable or disable the intrusion detection.

Block ICMP Requests, when this is enabled, the system will ignore any ping requests sent to it.

Intrusion Detection Settings Section

Failed Attempts Allowed, this is the number of attempts allowed before banning the IP address.

Find Time (seconds), this is the period that the number of failed attempts is placed in before getting banned.

Ban time (seconds), this is amount of time in which the possible intruder will be banned for. Set to -1 for permanent banning.

Send Notification, here you can define the email address to send an email notification to when an intrusion is detected.

Whitelist IP Addresses

WHITELIST IP ADDRESSES

BLACKLIST IP ADDRESSES

+ Whitelist an IP

Search:

IP/Domain	Description	Action
10.254.0.0/16	3Corp	 
10.11.0.0/16	3Corp	 

Showing 1 to 2 of 2 entries

Previous

1

Next

Host, IP Address or hostname.
Description, a brief description to identify the IP address.

Blacklist IP Addresses

+ Ban IP

Search:

IP Address	Jail	Date / Time	Action
No data available in table			

Showing 0 to 0 of 0 entries

Previous

Next

Here, you will find any IP address that has been banned. It will be shown on the table below. This table shows the IP Address that has been banned, as well as the Jail for the reason it was banned. If a host appears incorrectly on this list, you click on the  button to remove them from the list.

- Host**, Banned IP Address.
- Jail**, the type of action that was tried to be executed before the banning.
- , additional information about the banned IP address.
- , by pressing this option you can unban the IP Address.

8.3.2 Firewall Services

On this module you can create and modify the different firewall rules services.

If the Firewall is enabled, you can define various applications here. It is possible to add additional applications specific to your installation. To add an application, click on the “Add” button at the bottom of the module.

GENERAL

Show 25 entries

Search:

Service Name	Port	Protocol	Actions
PJSIP	50751	Both	 
DNS	53	Both	 
NTP	123	UDP	 
DHCP	67-68	UDP	 
HTTP	80	TCP	 
SSH	22	TCP	 
RTP	10000-20000	UDP	 
IAX2	4560	UDP	 
mDNS	5353	UDP	 
HTTPS	443	TCP	 
Asterisk HTTP Daemon	8088-8089	Both	 
SIP	5062-5063	Both	 
VPBX API	3500-3501	TCP	 
Vitxi WebRTC	8001	TCP	 

Showing 1 to 14 of 14 entries

Previous

1

Next

Name, here you can give the application a meaningful name to help you to easily recognize it. This name will be used to refer to the application in the Rules table.

Protocol, this determines the protocol that will be used by the application. Can be any one of TCP, UDP, or Both.

Port, the IP ports that are used by the application. This can be defined as a single port (e.g. 80), a range of ports (e.g. 10000:20000), or a list of ports separated by commas (e.g. 80, 10000:20000, 5060).

List of important ports

Service	Protocol	Port
SIP	UDP/TCP	5060
DNS	UDP/TCP	53
NTP	UDP	123
DHCP	UDP	67-68
HTTP	TCP	80
SSH	TCP	22
RTP	UDP	10000-20000
IAX2	UDP	4569
Sonata SwitchBoard	TCP	3001
Sonata SwitchBoard (HTTPS)	TCP	3008
mDNS	UDP	5353
HTTPS	TCP	443
Communications Systems HTTP Daemon	UDP/TCP	8088-8089
PJSIP	UDP/TCP	5062-5063
VPBX Dashboard	TCP	3000
VPBX Dashboard (HTTPS)	TCP	3005
MySQL	TCP	3306
AMI	TCP	5038
OpenVPN	UDP	1194

8.3.3 Firewall Rules

General

GENERAL

Show 25 entries

Search:

Service	Ports	Source	Destination	Action	
RTP	10000-20000			ACCEPT	 
PJSIP	50751			ACCEPT	 
HTTP	80			REJECT	 
HTTPS	443			ACCEPT	 
SSH	22	10.11.0.0/16	10.253.10.27	ACCEPT	 
DHCP	67-68			ACCEPT	 
DNS	53			ACCEPT	 
NTP	123			ACCEPT	 
IAX2	4569			REJECT	 
mDNS	5353		224.0.0.251	REJECT	 
SIP	5062-5063			ACCEPT	 
VPBX API	3500-3501			ACCEPT	 
Vitxi WebRTC	8001			ACCEPT	 
SSH	22	10.254.0.0/16	10.253.10.27	ACCEPT	 

Showing 1 to 14 of 14 entries

Previous

1

Next

- Service**, name of the services to use.
- Source**, this is used if you want the rule to be restricted to messages that are sent from a specific IP address or subnet only. Use any to allow all IP addresses. If you want to restrict the rule to a specific IP address, then you have to define the IP address with 32 as the subnet mask. If you use IPv6 then the subnet mask must be 128. In most cases you will define IP address as any.
- Destination**, here you can follow the same conventions as for Source above.
- Action**, configures whether this rule should Allow, Deny, or Reject access.
- **ACCEPT**, allow will pass all messages regardless of any rules that may be defined.
 - **REJECT**, reject means that the packet is not allowed to pass, and a response is given to the originator of the request.
 - **DROP**, drop means that the packet is discarded, but no response is given to the originator of the request.

8.3.4 DHCP Settings

Dynamic Host Configuration Protocol (DHCP) is the mechanism that dynamically allocates physical IP addresses to machines and devices on the network. You can choose to use an existing DHCP server on your network, or to use PBX as your DHCP server. If you want to use PBX as your DHCP server, check on the Enable button.

Be careful!



You can only have one active DHCP server on your network.

In this tab you will find the information about DHCP settings.

General

GENERAL

DHCP

EnabledDisabled

Disabled Interfaces

Start Address *

dhcpsettings.range_end *

Lease Time *

Days

Gateway

Primary DNS

Secondary DNS

NTP Server

Option 66

WINS

Static Leases

MAC Address	IP Address	Hostname (optional)
00:00:00:00:00:00	IP Address	Hostname

Add

DHCP, set this to enable if you want Attimo CloudPBX to act as a DHCP server for this network, or Disable if you already have a DHCP server on this network.

Disables Interface, you can disable DHCP on one or more interfaces. If you wish to specify multiple interfaces, separate them with commas. For example: eth0, eth0.201 **Start Address***, the first address on your network that can be allocated as a dynamic IP address.

End Address*, the last address on your network that can be allocated as a dynamic IP address.

Lease Time*, period that the DHCP server grants an IP address to a device. The device must renew its IP address before the end of the period.

Gateway, the default IP gateway address.

Primary DNS, domain Name System (DNS) translates Internet domain and host names to physical IP addresses in numerical notation, i.e. from mypbx.mydomain.com to 67.67.222.220.

Secondary DNS, here you can define a Secondary DNS to be used if your primary DNS fails to respond.

NTP Server, network Time Protocol (NTP) is a networking protocol to synchronize clocks between computer systems over the Internet.

Option 66, option 66 provides IP phones with an URL for configuration provisioning. Attimo CloudPBX Endpoint Manager provides the IP phones with configuration information in response to a HTTP request. The format of the request URL in this case looks like http://[pbx-ip-address]/xepm-provision/. If you define the PBX IP address or host name in the Option 66 field and check the 'Use for Endpoint Manager' checkbox then the correct format of the URL will be built

automatically. If you have already-prepared IP phone configuration files located in the /tftpboot directory, then you should put the PBX IP address or host name in the Option 66 field and uncheck the 'Use for Endpoint Manager' checkbox.

Use for End Point Manager, automatically format the Option 66 address for Endpoint Manager based on the address provided for Option 66 above. This will be done by prefixing http:// to the Option 66 address above and appending /xepm-provision/. For this to work correctly, the IP address provided above for Option 66 should only consist of the IP address or hostname of the server.

WINS, windows Internet Name Service (WINS) is a name resolution service that maps NetBIOS names to an IP address on the network that uses NetBIOS over TCP/IP (NetBT). The primary purpose of WINS is to support clients that run older versions of Windows and applications that use NetBIOS.

Static Leases Section

MAC Address, device MAC address.

IP Address, device IP address.

Host Name (optional), host name.

GEO Firewall

Increased security

With today's accessibility to the internet, the world has become even smaller, and we are able to communicate with anyone around the globe. We now have voice over IP connections established through the internet, and that makes our connections to our business and homes more reachable. But sometimes, people from places we do not intend to give access to our means of communication, in this case, our PBX systems, try to connect and use it for their own benefit.

At Attimo CloudPBX we want to make it easier to control who can connect or not to our PBX system. That is why we have created this brand-new commercial module called "Geo Firewall". With the Geo Firewall add-on, you can choose specific countries that can and cannot connect to your PBX system. And we have made it as easy as it gets.

GENERAL



How does it work?

To use the Geo Firewall add-on, you simply select the countries you wish to block access to your PBX, and then click "Save". It is as simple as that. The add-on will make sure to block any requests incoming from the blocked countries selected, and only allow from those that are allowed.

8.4 Add-ons

8.4.1 Add-ons

General

GENERAL

Addon	Installed Version	Available Version	Status	Actions
Sonata Billing	4.0.0-5	-		
Sonata Dialer	-	4.0.0-5		
Sonata Recordings	4.0.1-2	-		
Sonata Stats	4.0.2-1	-		
Sonata Switchboard	4.0.0-5	-		
System API	4.0.1-8	-		
Authentication Codes	-	4.0.0-4		
Branding	4.0.1-1	-		
Bulk Extensions	-	4.0.0-7		
Connect	-	4.0.0-7		
CRM Connector	-	4.0.0-1		
Custom Contexts	-	4.0.0-8		
Dynamic Destinations	4.0.0-8	-		
Geo Firewall	4.0.1-3	-		

Addon, Name of the Module or Application. **Installed Version**, Currently installed version. **Available Update**, Currently available version. **Status**, state of the Module or Application.

Buy License Activate/Buy the license, for this it is necessary to have bought a license in the Attimo CloudPBX store (<https://AttimoCloudPBX.org/en/AttimoCloudPBX-store/>),

See license with the possibility of revoke to be able to transfer it to another server.

Actions, there are three types of actions displayed depending on whether the plugin is installed or not:

Brief description of the plugin. Install add-ons.

Uninstall the add-ons.

Clean Cache, if you have any problems with the installations or updates maybe you need to clean the cache, please use this button.

Check Online, please use this button to check the latest updates.

Next, we list the available add-ons with a brief description of them:

- **Rebranding**, the rebranding add-ons is a simple but powerful module that allows you to customize the Attimo CloudPBX GUI according to your preferences (colors, logo, title, application name, etc.).

- **Bulk Extensions**, the Bulk Extensions add-ons is totally free and allows you to create massive extension ranges with common settings. It's a powerful time- saving tool when creating many extensions.
- **Communicator**, the Communicator add-ons allows you to centralize Attimo CloudPBX Communicator settings such as pauses and programmable keys. In addition, it allows you to create campaigns that can be used in the Communicator for automatic dialing (Outbound Campaigns).
- **Custom Context**, expand the power of Attimo CloudPBX with your own dial plan.
- **Dahdi**, this add-on installs the DAHDI drivers and all the necessary settings to be able to manage telephony devices on your PBX.
- **Domotic**, automate your environment with home automation and Communications Systems technologies.
- **Dynamic Destination**, allows you to make MySQL queries and API requests (HTTP / HTTPS), and depending on the response, the call will be routed to a specific destination according to the conditions defined in this module.
- **EPM**, this add-on installs all the necessary configurations to be able to provision various brands of telephones from the central.
- **Geo Firewall**, the Geo Firewall add-ons is very essential to prevent brute force attacks to your PBX from certain countries, therefore, installing this plugin will improve your security, decrease the likelihood of a hacker compromising your PBX and you will prevent VoIP fraud.
- **High Availability**, allows you to monitor and manage HA nodes from the GUI.
- **IVR Stats**, the IVR Stats add-ons allows you to get reports on your customers' interaction with your IVRs, allowing you to know which options are used the most or which departments are requested the most, and more.
- **Maintenance**, with the "Maintenance" add-ons you can easily clean your PBX and save a lot of storage: by deleting old recordings or converting recordings from wav format to mp3, you can also delete recordings with short durations and old CDRs.
- **Multi Tenant**, the multi-tenant add-ons allows you to manage multiple clients (tenants) with a single instance of Attimo CloudPBX, allowing you to significantly reduce operating costs.
- **Open VPN**, the OpenVPN add-ons is a commercial module and allows you to configure an OpenVPN server and its clients directly from the GUI. The free version comes with all the features, but you can only set up two clients at most.
- **Paging PRO**, this add-ons expands the paging functionalities, allowing you to schedule ads to page to the list of defined members. This is useful for scheduling school bells or announcements at companies, airlines, train stations, etc.
- **Phone Books**, the phone book add-ons are completely free and allows you to generate unlimited internal and external phone books on the fly and use those phone books in the different supported brands (Yealink, Grandstream, Alcatel, Fanvil, Xorcom, Htek).
- **Queues Callback**, Queued Callback allows you to optimize your customer experience by offering callers the ability to request a callback based on your call center conditions. When your call center experiences unexpected spikes in call volumes, the queued callback function steps in to offer your customer to call you back when an agent is available.
- **Task Manager**, the Task Manager add-ons is completely free and allows you to schedule tasks with custom user scripts through the GUI. The user must previously load the scripts to the path `/var/lib/Attimo CloudPBX/ scripts` and grant all the necessary permissions.

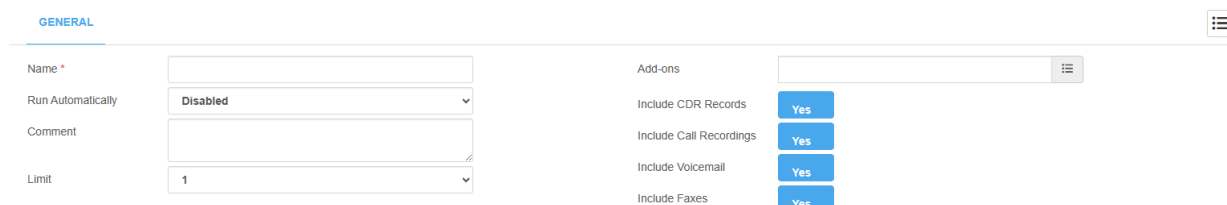
- **Trunk Passthrough**, this module allows you to send calls between trunks, thus turning your PBX into a recording server.
- **Sonata Switchboard**, advanced monitoring software for your PBX, take control of your PBX with this powerful tool. You can monitor queues, trunks, outgoing/incoming calls, perform actions on calls and many other functionalities. All information is in real time.
- **Sonata Recordings**, a recording management system for your PBX that allows you to label, qualify calls and everything related to call center supervision.
- **Sonata Billing**, is a powerful software to rate and manage your PBX calls, it provides a collection of reports to have total control of your calls.
- **Sonata Stats**, a system through which you can obtain detailed and summarized reports of all the activity in your call center. Among the most important reports we have the SLA, Unanswered Calls, etc.

8.5 Tools

8.5.1 Backup & Restore

In this tab you will find the information about Backup & Restore.

General



The screenshot shows the 'General' tab for the Backup & Restore settings. It features a form with the following fields and options:

- Name ***: A text input field.
- Run Automatically**: A dropdown menu currently set to 'Disabled'.
- Comment**: A text area for adding a comment.
- Limit**: A dropdown menu currently set to '1'.
- Add-ons**: A section with a search bar and a list of checkboxes:
 - Include CDR Records**: Yes
 - Include Call Recordings**: Yes
 - Include Voicemail**: Yes
 - Include Faxes**: Yes

Name, Descriptive name for this backup.

Run Automatically, it allows you to use a Cron previously created at PBX/Tools/Cron Profiles to automatically create a backup with your selected settings.

Comment, a user-defined comment that will be added to the backup.

Limit, it allows you to define the number of backup copies that will be stored. When the limit is reached, the oldest copy will be deleted.

Include CRD Records, enabling this will add CDR records to the backup. **Include Call Recordings**, enabling this will add call recordings to the backup. **Include Voicemail**, enabling this will add voicemail boxes to the backup.

Backup List

Date & Time, date and time the backup was made Backup, name of the file that contains the backup. Attimo CloudPBX Version, Attimo CloudPBX version of the backup.

Actions, possible actions that can be performed with the backup are:

-  Download
-  Restore
-  Delete

8.5.2 Maintenance

This is a simple add-on with powerful settings that allows you to save space in your PBX. This is a commercial add-on and you may buy it through the following link: <https://attimotelecom.com.br/produtos>

How it works.

The maintenance add-on has various settings, between them, the possibility to assign a Cron Profile item to schedule the execution of the configured options. These are the most powerful settings:

GENERAL

Tenant	Attimo	Schedule	None
Delete Oldest CDR		Audio Quality	Good (16k)
Delete Oldest Recordings	180	Convert Recordings	No
Delete Oldest Voicemails	60	Enabled	No
Delete Short Recordings	1		

Tenant, allows you to select the tenant on which the maintenance configurations will be applied.

Clear Oldest CDR, allows you to define the maximum number of days that CDR should be retained. The CDR with more days than defined here will be deleted.

Clear Oldest Recordings, this option allows you to define the maximum number of days that recordings should be retained, allowing you to keep only the most recent recordings.

Clear Short Recordings, allows you to define the minimum duration in seconds for a recording to be considered as too short, and delete it.

Schedule, it allows you to define the schedule in which the maintenance of the PBX will be executed (Conversion of Recordings, Cleaning of Recordings and CDR, etc). If no schedule is selected, all the maintenance options will be disabled.

Convert Recordings, this option allows you to enable the conversion of CDR recordings from WAV to MP3.

Enabled, it allows you to enable or disable the PBX maintenance.

8.5.3 Branding

This simple but very useful add-on allows you to customize the Attimo CloudPBX colors, logos (Mobile and Desktop Version), browser title and others.

GENERAL GUI LOGIN DESIGN ADDONS

Base Color	#49a8eb	Login Footer	Attimo<a> is a registered trademark of 3Corp Technology.
APP Title	Attimo	Meet URL	
Application Name	Attimo	Disable Branding	No
Slogan	Communication as a service.	Desktop Logo (Dark Mode)	
Facebook Page	3CORPTechnology SA		
Instagram Page	3corptechnology		
Twitter Page	3corp		
YouTube Channel	user3CORPTechnology		
Desktop Logo			
	Change Remove		Change Remove
Mobile Logo			
	Change Remove		Change Remove

Base Color, main color used for active menu items, active form tabs and selected text.

APP Title, allows you to customize the header title that appears in the browser tab.

APP Name, it allows you to modify the main Tenant name.

Slogan, it allows you to customize the Slogan that appears in the login screen.

Facebook Page, it allows you to customize the Facebook Page link that appears in the login footer.

Instagram Page, it allows you to customize the **Instagram Page** link that appears in the login footer.

Twitter Page, it allows you to customize the Twitter Page link that appears in the login footer.

YouTube Channel, it allows you to customize the YouTube Channel link that appears in the login footer.

Log In Footer, it allows you to customize the footer content in the login screen.

Meet URL, URL to be used by default for conferences generated through the video conferences module.

Desktop Logo, logo to show when accessed from a desktop computer.

Mobile Logo, logo to show when it is accessed from a mobile.

Login Design

GENERAL GUI **LOGIN DESIGN** ADDONS

Background Color	#ffffff	Submit Button Color	#49a8eb
Font Color	#49a8eb	Add-ons Menu	#49a8eb
Form Background	#b3cce7	Disable Background Image	Yes

Background Image



Select Image

Logo



Change Remove

Login Background, background color for the login page. If left in blank, the default background will be used.

Font Color, font color for the login page. If left in blank, the default color will be used.

Form Background, background color for left panel in the login form.

Submit Button Color, background color for the login button.

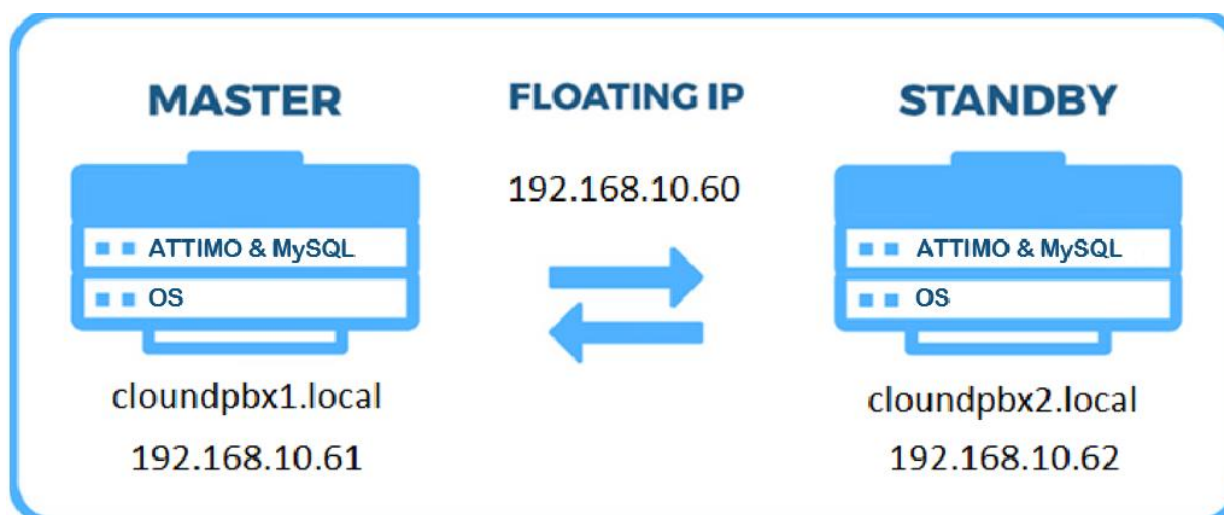
Add-ons Menu, background color for the add-on's menu button.

9. High Availability

High availability is a characteristic of a system which aims to ensure an agreed level of operational performance, usually uptime, for a higher-than-normal period.

Make a high-availability cluster out of any pair of Attimo CloudPBX servers. Attimo CloudPBX can detect a range of failures on one Attimo CloudPBX server and automatically transfer control to the other server, resulting in a telephony environment with minimal down time.

- At the time of installation leave the largest amount of space on the hard drive to store the variable data on both servers.
- If you are going to restore some Backup from another server that is not in HA, restore it first in the Master server before creating the HA. This should be done as the backup does not contain the firewall rules for HA to work.
- The Attimo CloudPBX team does not provide support for systems in an HA environment because it is not possible to determine the environment where it has been installed.
- We recommend that Server 2 is completely clean if nothing is installed, otherwise the Script could give errors and the process would not be completed.





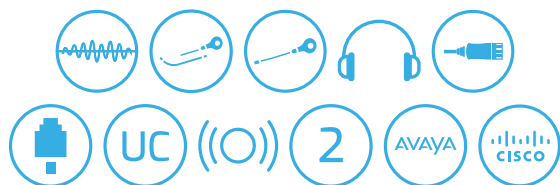
Epko series

Preferida entre os maiores call centers no Brasil, a linha de Headsets Epko tornou-se referência no mercado de headsets profissionais aliando durabilidade incomparável, conforto excepcional, design inovador e som de incrível fidelidade. Desenvolvidos especialmente para utilização contínua em call centers e por profissionais que necessitam de um headset de alta performance.

Epko series



Epko Compact



ESPECIFICAÇÕES TÉCNICAS

Peso	✓ Monoauricular 51 gramas (sem cabo) ✓ Biauricular 71 gramas (sem cabo)
Auricular	✓ 41 mm (sem espuma)
Versões	✓ Monoauricular ✓ Biauricular ✓ RJ9 ✓ QD - Quick Disconnect
Tecnologia	✓ Compatível com sistemas telefônicos digitais, softphones (VoIP) e analógicos
Microfone	✓ Microfone de Eletreto ✓ Omnidirecional (Compact e Voice Guide) ✓ Noise Cancelling (Noise Cancelling)
Sensibilidade do microfone	✓ -43dBV (Speech Bandwidth) 100Hz ~ 10kHz (Noise Cancelling) ✓ -54dBV (Speech Bandwidth) 100Hz ~ 6,8 kHz (VG e Compact)
Cápsula Receptora	✓ Cápsula Dinâmica de 27 mm ✓ Membrana Mylar ✓ Balanceado de 10 Hz a 10 kHz ✓ Impedância 150 ohms +/- 10% @ 1 KHz ✓ Sensibilidade -90 dB (Maximum output 106 dB)
Garantia	✓ 2 anos

PRINCIPAIS CARACTERÍSTICAS

Materiais e Acabamento Premium

• Polímeros de engenharia de última geração e peças em aço inoxidável garantem alta durabilidade, performance e sofisticação.

• A bucha reprojetada e o cabo reforçado com filamentos de Kevlar® proporcionam uma combinação perfeita de flexibilidade e resistência.

Conforto e Ergonomia

• Avançado sistema ergonômico com tiara de aço inoxidável revestida em material hipoalergênico, apoio lateral emborrachado e protetor auricular em espuma proporcionam o máximo de conforto e estabilidade para uso prolongado.

Qualidade de Áudio de Alta Definição HD

• Sofisticado sistema eletrônico de cancelamento de ruídos, proteção contra choques acústicos e equalização de áudio garantem ligações cristalinas e total segurança ao usuário, seguindo normas da Anatel, IEEE (Institute of Electrical and Electronics Engineers) e EC (European Community).

• Projetado de acordo com as novas tendências em Comunicações Unificadas. Na versão com cabo Quick Disconnect (QD), o mesmo headset Epko pode ser conectado à terminais telefônicos com cabo RJ9, ou à PC's e notebooks com Interface USB Felitron.

• Disponível nas versões com microfone cancelador de ruídos, tubo de voz substituível e tubo de voz fixo.

VERSÕES



Versão sem QD

- Conecte o headset diretamente em terminais telefônicos digitais e VoIP.
Consulte Guia de Compatibilidade

Produtos	PN
Epko Noise Cancelling	01114-9
Epko Noise Cancelling Biauricular	01116-1
Epko Voice Guide	01113-1
Epko Voice Guide Biauricular	01114-5
Epko Compact	01113-7
Epko Compact Biauricular	01113-9

COMPATIBILIDADE



Versão com QD

- Conecte a Interface USB ou a Interface USB Slim ao headset e utilize-o em aplicações VoIP em seu PC ou notebook.
- Conecte o headset ao Cabo QD Leve para utilização em terminais telefônicos digitais.
Consulte Guia de Compatibilidade



Cabo QD Leve (RJ9)



Interface USB

Produtos	PN
Epko Noise Cancelling QD	01114-0
Epko Noise Cancelling Biauricular QD	01116-2
Epko Voice Guide QD	01113-2
Epko Voice Guide Biauricular QD	01114-6
Epko Compact QD	01113-8
Epko Compact Biauricular QD	01113-0
Interface USB	01130-1
Interface USB Slim	01131-2
Cabo QD Leve	01210-6

Epko

Epko Compact

[Compact](#)

[Monoauricular](#)

[RJ](#)



COMPRAR

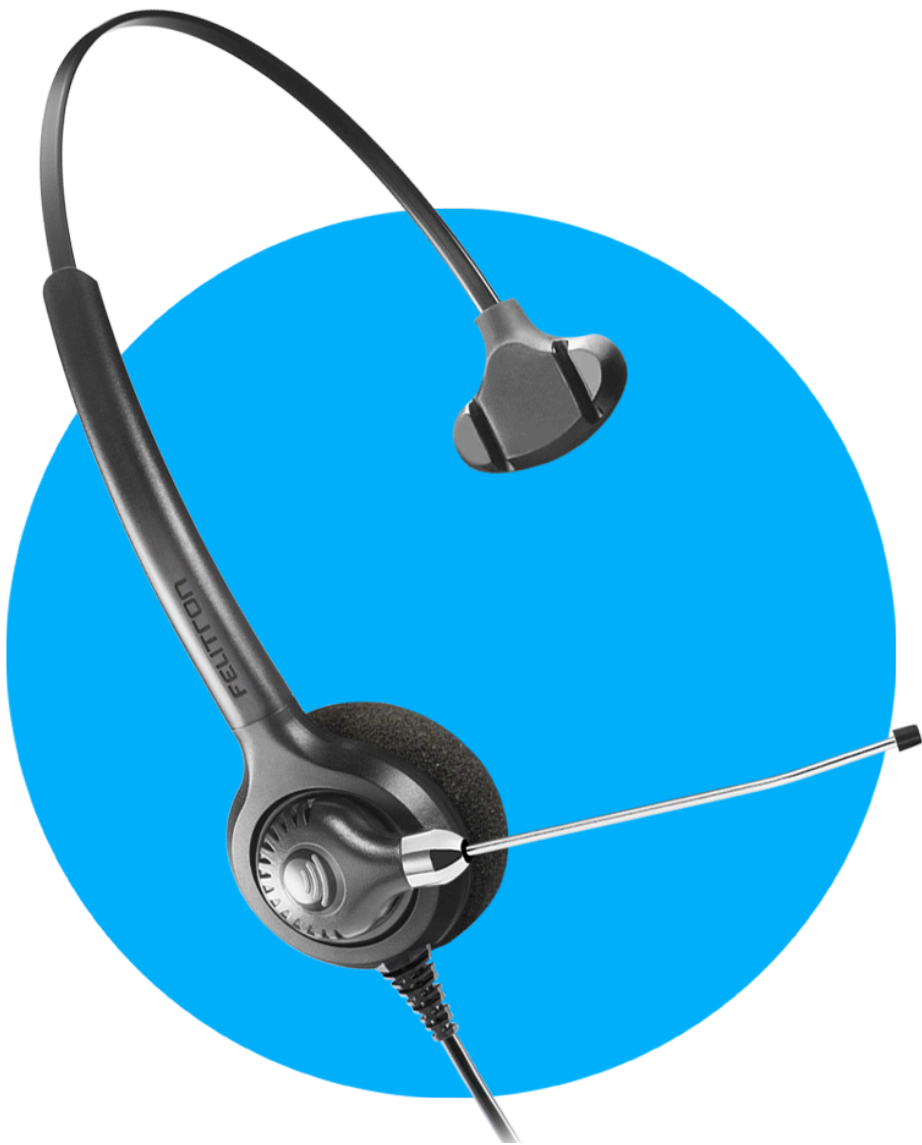
Sobre o produto



VISÃO GERAL

Preferida entre os maiores call centers no Brasil, a linha de headsets Epko tornou-se referência no mercado de headsets profissionais aliando durabilidade incomparável, conforto excepcional, design inovador e som de incrível fidelidade, desenvolvidos especialmente para utilização contínua em call centers e profissionais que necessitam de um headset de alta performance.

Características



- Tubo de voz fixo;
- Protetor auricular em espuma de alta densidade;
- Cabo super flexível de alta durabilidade reforçado com Kevlar®;
- Tiara revestida com material hipoalergênico e apoio lateral emborrachado;
- Conjunto de voz em aço inoxidável com ajuste preciso de posicionamento;
- Sistema exclusivo de condicionamento de som e redução de ruídos ambiente;
- Haste do microfone em material flexível e de alta durabilidade;
- Proteção eletrônica contra ruídos e choques acústicos;
- Indicado para call centers que utilizam tecnologia digital e VoIP;
- Dispensa o uso de amplificadores e adaptadores externos;
- Avançado sistema de recepção com excelente performance acústica;
- Conector RJ9.



Conforto



Durabilidade



Performance



Acessórios



Protetor Auricular



Suporte de Monitor



Cabos

CONHEÇA OUTROS MODELOS



EPKO VOICE GUIDE VOIP



EPKO NOISE CANCELLING



EPKO COMPACT VOIP

INSCREVA-SE PARA RECEBER NOTÍCIAS E OFERTAS EXCLUSIVAS DA FELITRON

digite seu e-mail...PARTICIPE

Suporte ao Cliente
Guia de Compatibilidade
Onde comprar

Home
Quem somos
Blog
FLT Lab
Contato

Choose your Language



Área Restrita
Acesso exclusivo para parceiros autorizados

SAC Felitron - 55 11 4096 5001
Whatsapp - [55 11 4096 5001](#)
[Fale pelo nosso Chat](#)
[E-mail: faleconosco@felitron.com.br](mailto:faleconosco@felitron.com.br)

Horário de atendimento
Segunda a sexta das 9h às 17h

Escritório Comercial
Rua Enxovia, 472 - Conj. 1115
CEP: 04711-030 - São Paulo - SP

Fábrica
Rua Antônio da Chagas, 723 - São Paulo - SP
Brasil - Cep. 04714-001

Alianças Estratégicas



Media gateway com interfaces modulares e SBC



Principais características

- Composição modular: Até 16 módulos externos para compor com as tecnologias de E1/T1, FXS, FXO ou GSM
- SBC integrado com até 2000 sessões VoIP
- Até 2000 canais TDM (até 64 E1's)
- Suporte SS7/SIGTRAN e SIP-I
- Suporte à classificação de chamadas
- Sobrevivência e Autorização de Registros
- Alta disponibilidade em modo ativo/inativo

Aplicações

- Indicado para cenários robustos, de grande capacidade de chamadas, e que necessitam de gerenciamento completo da operação de telefonia
- Classificação de chamadas e roteamento inteligente
- Gateway TDM – VoIP de grande capacidade com até 64 E1's
- SBC integrado com suporte à Autorização de Registros
- Interconexão com suporte a SS7/SIGTRAN e SIP-I
- Controle de custo da fatura telefônica e fidelização de operadoras de longa distância
- Gestão inteligente e centralizada

Visão geral

O KMG 3200 One, é um produto da linha de Media Gateways da Khomp. Um dispositivo de alta densidade de chamadas simultâneas, com suporte para até 64 links E1/T1, ou 2000 canais TDM, que podem ser utilizados também pelas tecnologias GSM, FXO e/ou FXS. É ideal para estruturas de rede confiáveis e que necessitam de máxima qualidade de voz.

O KMG 3200 One possui 13 portas de rede que podem ser usadas na interconexão entre redes distintas e/ou na conexão dos módulos de telefonia.

Os módulos de telefonia podem ser cascadeados entre si, permitindo a quantidade máxima de 16 módulos.

Possui recursos avançados de roteamento e segurança SBC (do tipo B2BUA). Conta ainda com recursos de classificação de chamadas, sobrevivência local, alta disponibilidade (HA, High Availability) e monitoramento inteligente dos canais em tempo real.

Capacidade de chamadas

O KMG 3200 One. possui capacidade de 2000 chamadas simultâneas, sejam elas TDM ou VoIP.

No caso de uso de transcoding do codec padrão VoIP G.711 para os codecs G.729 e G.722, essa capacidade total cai pela metade, ou seja, 1000 chamadas simultâneas para qualquer tecnologia (Any-to-Any).

Para as chamadas VoIP, existe ainda a opção de configuração em modo Bridge*, com capacidade de 2000 chamadas simultâneas, e com a vantagem de poder utilizar qualquer codec de áudio ou vídeo.

Roteamento de chamadas

Tenha maior controle nos gastos com tarifas de telefonia, através da configuração de rotas por prefixo ou pela fidelização da operadora de telefonia, o que possibilita direcionar as chamadas para as operadoras que oferecem o melhor custo benefício para cada ligação, proporcionando custos menores nas tarifas.

Cadastre o roteamento de chamadas com transbordo automático por horários ou retry, ordene as rotas por prioridade e modifique os números de A e de B (se necessário), possibilitando assim uma enorme variedade de combinações, incluindo a criação de rotas de menor custo, contingência e balanceamento.

O failover de rotas é outro recurso importante para quem não pode ficar sem os serviços de telefonia em sua rede. É implementado utilizando as rotas juntamente com a monitoração do servidor VoIP de destino da chamada, e se o servidor VoIP não responde aos comandos enviados pelo monitoramento, o KMG ignora a rota e busca por outra rota compatível.

Além disso, utilize scripts de roteamento para facilitar a adequação a diversos cenários. Todas as informações de roteamento podem ser armazenadas e disponibilizadas para análise através dos arquivos de CDR, gerados pelo KMG 3200 One, com formato customizado e suporte a RADIUS.

Interconexão com suporte SS7/SIGTRAN e SIP-I

Múltiplas possibilidades de interconexão, através de Suporte a Ponto de Sinalização e Ponto de Transferência de Sinalização (PS e PTS), com sinalização SS7 e SIGTRAN. Além do suporte ao protocolo SIP-I, possibilitando novos cenários de expansão sem a necessidade da preocupação com os enlaces TDM. Desta forma o KMG 3200 One torna-se o equipamento ideal para operadoras, abrangendo diversos cenários com possibilidade de expansão futura.

Módulos de telefonia (itens opcionais)

O KMG 3200 ONE possui a característica de modularização, o que possibilita que ele seja montado de acordo com o modelo de negócio que será aplicado, aceitando simultaneamente as interfaces E1/T1, FXS, FXO e GSM. Veja mais detalhes sobre os módulos externo de telefonia:

- **Módulo KMG GSM – 160 (H – para 3G):** Module for applications that need GSM channels and advanced voice resources. This module has up to 16 GSM channels with a 3G six-band GSM interface with fallback to 2G, with 2 SIM cards per channel, one active and one in stand-by, in addition to 16 SIP channels for VoIP.
- **Módulo KMG FXS 240:** Módulo para aplicações que necessitam de interface de ramal analógico. Este módulo possui 24 canais analógicos FXS e 24 canais SIP para VoIP além de protocolos de PBX como transferência, segunda linha e pêndulo.

- **Módulo KMG FXO 120:** Módulo para aplicações que necessitam de entroncamento analógico. Este módulo pode ter 4, 8 ou 12 canais analógicos FXO, sendo 1 canal SIP para cada canal analógico para VoIP. Possui protocolos de PBX sendo possível ter geração e detecção de Flash.
- **Módulo KMG Modular:** Módulo que integra as interfaces GSM, FXS, FXO, E1/T1 e VoIP em um único hardware. As interfaces podem ser adquiridas conforme a necessidade da aplicação, podendo combinar 3 das seguintes interfaces: 1 ou 2 links E1/T1, 8 canais FXS, 4 canais FXO, 1 ou 2 canais GSM. Cada interface possui as mesmas características de desempenho e funcionalidades dos módulos descritos acima, porém combinados em um único equipamento.

Para mais opções modulares, consulte o manual do produto.

E1/T1 Bypass para segurança da solução (item opcional)

O E1/T1 Bypass proporciona contingência para os produtos com estes links. Instalado dentro do próprio equipamento, comuta fisicamente o link 1 ao 2, fazendo a transferência de um link E1/T1 para outro caso ocorra falhas do servidor.

Monitoramento de chamadas: Insight! (item opcional)

O Insight! aplica o monitoramento efetivo em dashboard, em tempo real, com gerenciamento inteligente das chamadas realizadas pelo Gateway, como a quantidade de chamadas, o tempo médio das ligações, as causas de desligamento, além da geração de alertas baseados em parâmetros pré-definidos para manter elevado o rendimento da operação.

Sobrevivência - SAS (item opcional)

A sobrevivência, (SAS – Stand Alone Survivability) assegura a continuidade da comunicação de telefonia caso o PBX IP fique indisponível. O KMG 3200 One, com licença de sobrevivência aplicada, assume as funções básicas do PBX IP, como por exemplo: realização e recebimento de chamadas entre ramais, ligações externas, e transferência de chamadas. Desta maneira, a comunicação não fica comprometida até que o PBX IP esteja disponível novamente.

Alta Disponibilidade (item opcional)

O KMG 3200 One possui sistema integrado de alta disponibilidade, baseado no conceito de equipamento ativo/inativo (1+1), com replicação automática das configurações que permite, em caso de falha do equipamento ativo, a troca automática para o equipamento inativo, o qual irá assumir os endereços de rede e tarefas de roteamento, sendo promovido a ativo, evitando paradas prolongadas na operação devido a falhas de hardware, troca ou manutenção do gateway ativo.

SIP trunking (item opcional)

Através do KMG 3200 One é possível realizar chamadas para conexão SIP. Esse tipo de solução é ideal para empresas e instituições com grande necessidade de comunicação através de centrais telefônicas IP e que procuram qualidade de serviço, flexibilidade e custos acessíveis nos serviços de voz.

O KMG 3200 One, possui 3 modos de operação VoIP: No modo G.711, permite até 2000 chamadas VoIP. Em caso do modo transcode a capacidade máxima é de 1000 chamadas VoIP. Já no modo bridge a capacidade máxima é de 2000 chamadas VoIP, com a vantagem de poder ser utilizado qualquer codec de áudio ou vídeo.

Com isto, diversos outros recursos de SBC e segurança são agregados ao equipamento, permitindo interoperabilidade entre redes e protocolos utilizando suas 13 interfaces de rede, bem como travessia NAT e demais recursos fornecidos através da Autorização de Registros (licença separada).

Saiba mais sobre os recursos do SBC da Khomp com nossos consultores comerciais.

Imagens do produto



Legenda: Visão frontal do KMG 3200 One.



Legenda: Visão traseira do KMG 3200 One.

Especificações técnicas



- O hardware do produto pode ser substituído sem aviso prévio.
- A substituição acontece quando a matéria prima não é encontrada no mercado ou quando surgem hardwares melhores.
- Quando o hardware é substituído, o produto vai operar com o mesmo potencial da configuração anterior.

Suporte a troncos E1/T1

- Canais de rede: até 64 links E1/T1
- Protocolos de rede: ISDN e R2 digital. É possível configurar protocolos diferentes em cada um dos links.
- Protocolos de PBX: EL7, Line Side, LC e QSIG (SSCT e CT)
- Opções de conector:
 - BNC coaxial (75 Ohms)
 - RJ45 (120 Ohms)
- 30 canais SIP para cada link E1/T1 (G.711)
- Suporte a Ponto de Sinalização e Ponto de Transferência de Sinalização (PS e PTS) em SS7 e SIGTRAN (Licenciamento opcional)
- Suporte SIP-I

Status do sistema

- Status do sistema via web
- Status dos troncos e canais via Web
- Diagnóstico detalhado dos links E1/T1
- Suporte SNMP

Interfaces de operação

- Configuração, monitoração, administração e diagnóstico via Web
- Controle de acesso e registro de alterações por usuário na Interface Web
- Geração de logs de sinalização e de sistema
- Análise de log de chamada integrado na interface (R2/ISDN)
- Captura de pacotes via Web
- Interface serial (RS-232 conector DB-9)

Controle de tráfego

- Limitação de chamadas simultâneas por rede

Codecs suportados

- G.711 A-law e μ -law, nativo no sistema, para todas interfaces
- G.729A, G722, GSM, DVI, T-38; em transcoding
- VoIP bridge para qualquer codec, inclusive codecs de vídeo (sem suporte à classificação de chamadas)

Roteamento de chamadas

- Roteamento de chamadas LCR – roteamento de menor custo
- Roteamento baseado em origem, destino, horário e priorização
- Fidelizador de rota (capacidade de mudar o número de destino)
- Consulta base de portabilidade
- Fallback para rotas em falha
- Failover retry baseado nas causas de falhas
- Roteamento por script
- Balanceamento de carga
- Perfil de rotas
- Até 500 tentativas de chamadas por segundo (CAPS)
- Até 2000 registros simultâneos (Recurso compartilhado entre Sobrevivência e Autorização de Registros)

Recursos VoIP

- Manipulação de número de destino (to) e número de origem (from)
- Monitoração de NAPS (pontos de acesso à rede) ou Keep Alive (envia pacotes UDP ao roteador para indicar que a porta está em uso, sem afetar a largura de banda)
- Fallback de proxy SIP
- Seleção de modo de envio DTMF: In band, Out band – RTP (RFC 2833) ou Out band – SIP Info
- Adição, remoção e retransmissão de cabeçalhos
- Transcoding (conversão entre os codecs G.711, G.729 e G.722)

Sobrevivência - SAS

- Encaminhamento de chamadas entrantes e saídes
- Transferência com e sem consulta
- Fallback automático de proxy

QoS (Controle de Qualidade)

- DiffServ - RFC 4594 4 (classificação e gerenciamento de tráfego)
- VLAN Tagging

Registro de chamadas

- Geração de CDR em formato customizável
- Monitoração de ocupação dos canais
- Contadores de ligações por canal
- Opções de download em arquivo CSV (compatível com Microsoft Excel)
- Exportação automática por FTP
- Utilização do protocolo RADIUS para efetuar o Accounting (bilhetagem)

Interoperação

- Interoperação de Fax (T.38 com fallback para G.711)
- IPv4 para IPv6
- RTP com conversão entre UDP, TCP e SRTP (SDPS e DTLS)
- Entroncamento SIP
- Roteamento direto do Microsoft Teams. Fase beta, interoperabilidade com e sem bypass de mídia.

Módulos de telefonia

FXS

- Canais de rede: 24 canais analógicos FXS
- Protocolos de PBX: transferência, segunda linha, pêndulo e conferência
- Cadências de ring configuráveis
- Compatibilidade com FOP (Flash Operator Panel)

FXO

- Canais de rede: 4, 8 ou 12 canais analógicos
- Modularidade: 3 x 4 linhas
- Protocolos de PBX: geração e detecção de flash
- Impedância de linha configurável para 900 ou 600 Ohms

GSM

- Modular até 16 interfaces GSM
- Capacidade para 2 SIM card por canal, um ativo e outro em stand-by
- Permite diferentes operadoras no mesmo módulo
- 3G Six Band: 800/850/900/1700/1900/2100 MHz
- 2G Quad Band: 850/900/1800/1900 MHz
- Tamanho do SIM card: mini-SIM (2FF)

Garantias e certificações

- Garantia total (legal + garantia Khomp): 1 ano
 - Garantia legal: 90 dias
 - Garantia Khomp: 9 meses
- Certificação Anatel
- Indústria certificada ISO 9001

Travessia NAT

- Interligação de diferentes redes
- Configuração de IP externo
- STUN

Segurança

- Acesso através do protocolo HTTP ou HTTPS
- Prevenção de fraudes: bloqueio de chamadas por destino e origem
- Proteção DoS/DDoS
- Ocultação de topologia de rede
- Protocolos SIP TLS e SRTP (SDPS e DTLS)
- Controle de acesso – ACL (lista de permissões e lista de bloqueios)
- Proteção contra pacotes mal formados
- Register authorization*

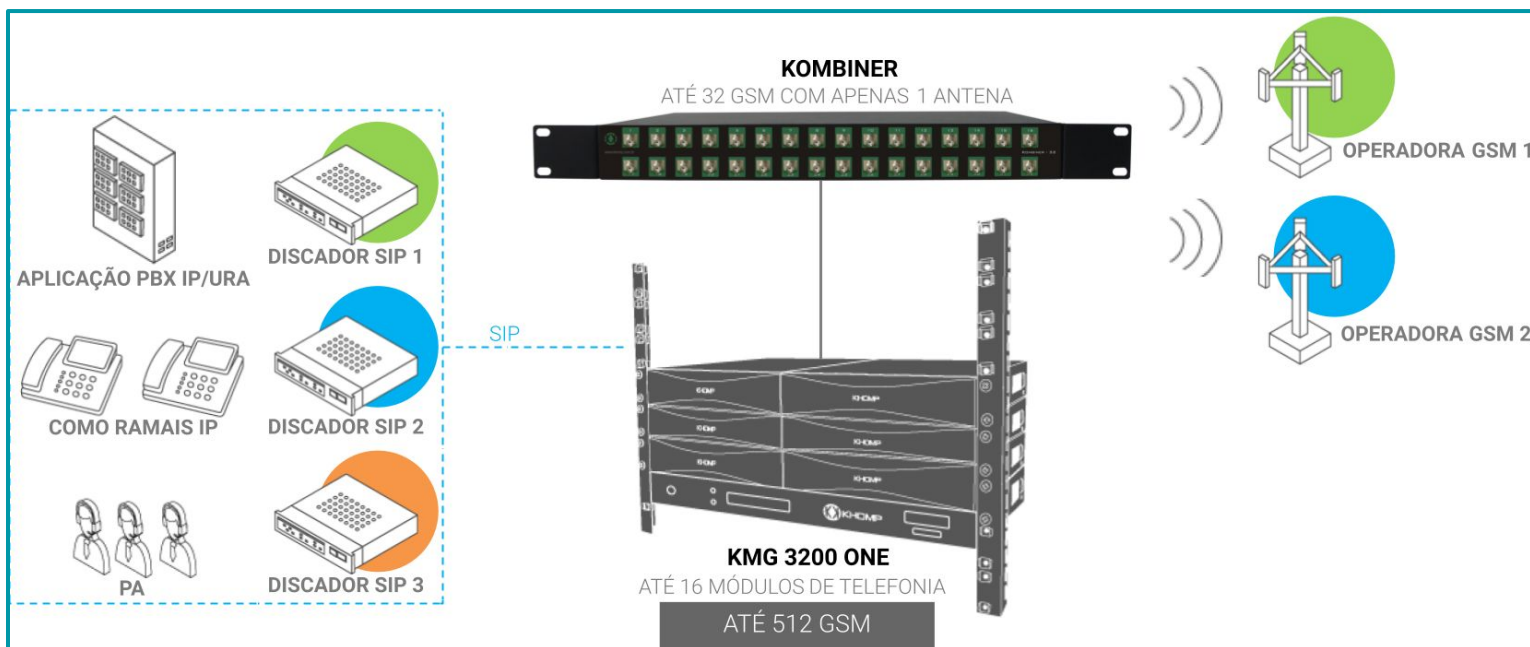
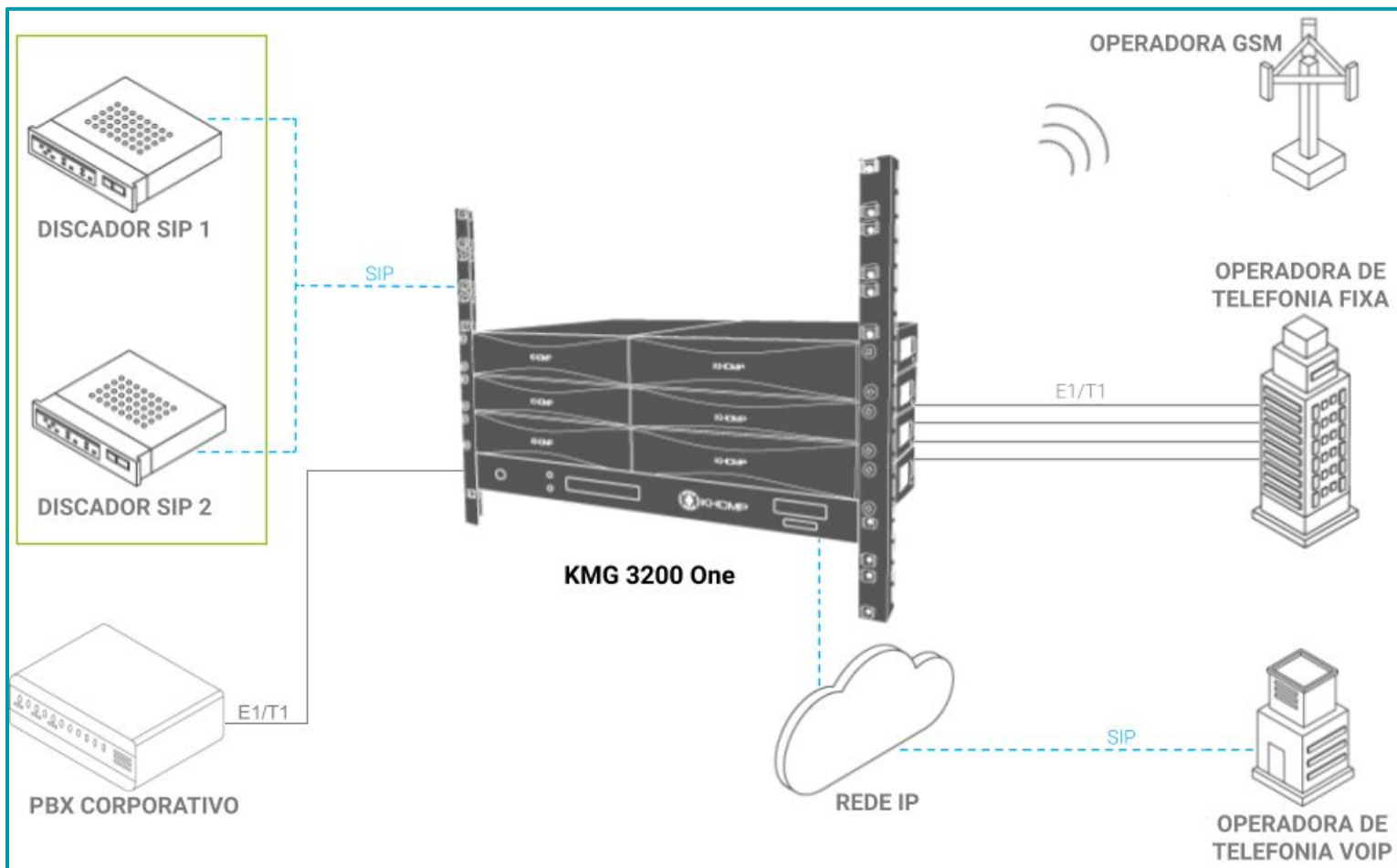
Outras funcionalidades

- Provisionamento (exportação e importação de configurações)
- Histórico e restauração de alterações de configuração via Interface Web
- Terminal remoto com advanced CLI (Command Line Interface)
- Suporte a TR-069
- Suporte as normas ITU-T, G.165 e G.168
- Alta disponibilidade (1+1)
- Tratamento de sinalização acústica realizado por hardware através de DSPs
- Detecção automática de tom de fax (2100 Hz) habilitando automaticamente o cancelamento de eco

Físico/Ambiental

- Fonte de energia redundante
 - Entrada: 110–240 VAC, 50/60 Hz
 - Consumo máximo de energia: 150 W
- 13 interfaces de rede gigabit 10/100/1000 Mbps
- Display OLED 0.96" (1,90x2,73 cm)
- Módulo padrão 1U para rack de 19"
- Módulos de telefonia utilizam 1U para cada 2 módulos
- Dimensões: 437,8x44,45x380 mm
- Peso aproximado (sem embalagem): 7,5 Kg

Modelos de aplicação



PROTETOR auricular espuma

Sobre o produto

▼

[COMPRAR](#)

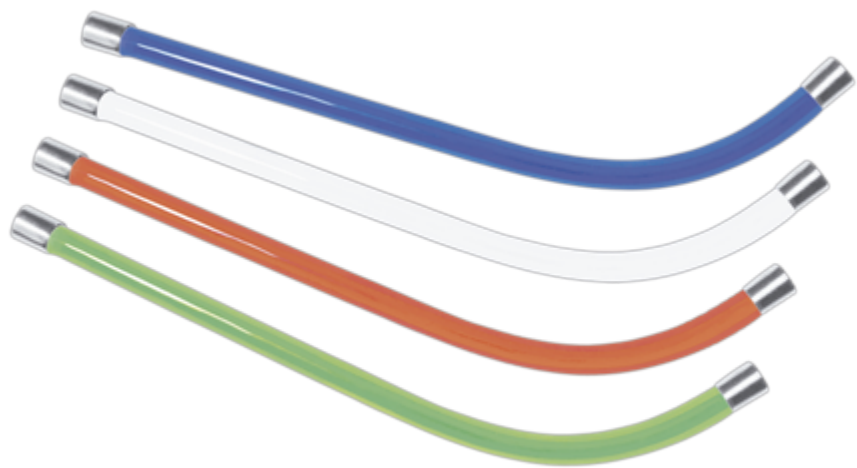


Descrição ▼

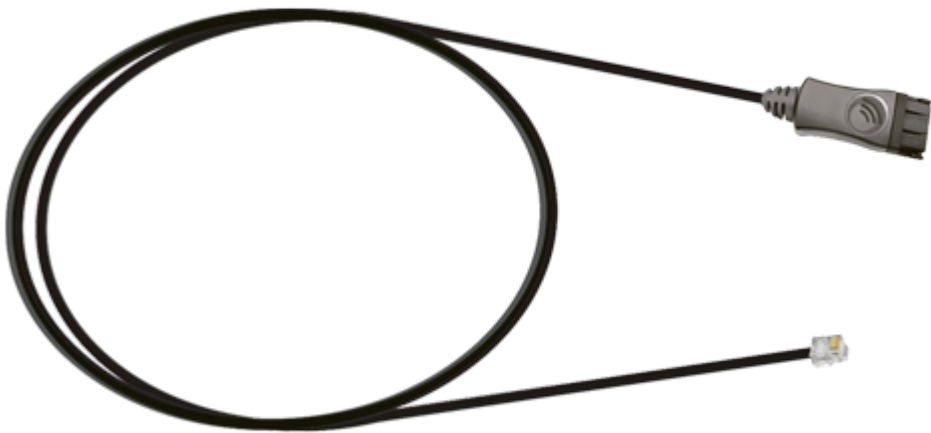
Características ▲

- Material de alta durabilidade e hipoalergênico
- Espuma de Poliuretano
- Tipo filtral
- Duplo corte interno que garante níveis dentro do limite e normas vigentes e mais conforto ao usuário
- Compatível com a linha Epko series

OUTROS ACESSÓRIOS



Tubo de Voz



Cabos



Suporte de Monitor

INSCREVA-SE PARA RECEBER NOTÍCIAS E OFERTAS EXCLUSIVAS DA FELITRON

digite seu e-mail...

PARTICIPE

Suporte ao Cliente
Guia de Compatibilidade
Onde comprar

Home
Quem somos
Blog
FLT Lab
Contato

Choose your Language



Área Restrita

Acesso exclusivo para parceiros autorizados

SAC Felitron - 55 11 4096 5001
Whatsapp - [55 11 4096 5001](#)
[Fale pelo nosso Chat](#)
[E-mail: faleconosco@felitron.com.br](mailto:faleconosco@felitron.com.br)

Horário de atendimento
Segunda a sexta das 9h às 17h

Escritório Comercial
Rua Enxovia, 472 - Conj. 1115
CEP: 04711-030 - São Paulo - SP

Fábrica
Rua Antônio da Chagas, 723 - São Paulo - SP
Brasil - Cep. 04714-001

Alianças Estratégicas





Recordings

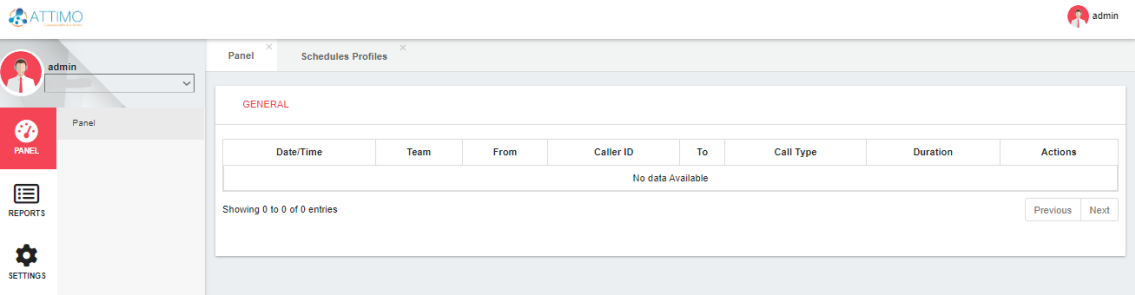
INDEX

Panel.....	3
Reports	5
Settings	7
Installation	12

Panel

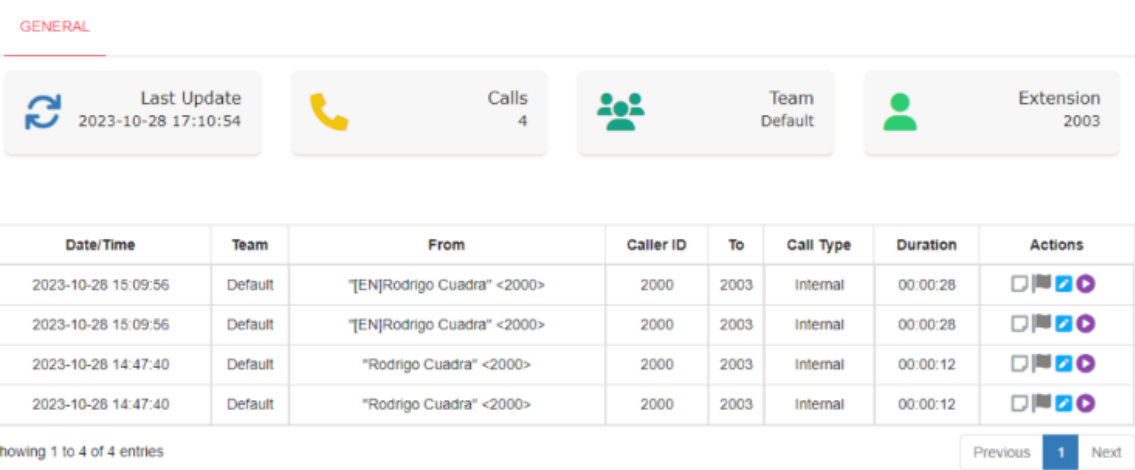
Admin Panel

The Manager/Supervisor/Administrator Panel displays the latest calls or recordings with the ability to set flags, create notes, set a reference or listen to the recording.



Agent Panel

The Agent Panel displays the latest calls or recordings with the ability to set flags, create notes, set a reference or listen to the recordings.



We can also observe the last update of the calls that we are seeing with the possibility to press the button to update.

Manager/Supervisor/Administrator Panel

The Manager/Supervisor/Administrator Panel displays the latest calls or recordings with the ability to set flags, create notes, set a reference or listen to the recording.

Show

50

 entries

CSV

Save

Date/Time	Team	Caller ID	From	To	Call Type	Duration	Rating	Actions
2024-04-09 17:16:50	Default	"<6137740250>	6137740250	1996	Incoming	00:03:47	☆☆☆☆☆	
2024-04-09 17:16:50	Default	"<6137740250>	1996	6133291996	Outgoing	00:03:47	☆☆☆☆☆	
2024-04-04 15:12:53	Default	"<21991190858>	1959	6133291959	Outgoing	00:00:51	☆☆☆☆☆	
2024-04-04 15:12:52	Default	"<21991190858>	21991190858	1959	Incoming	00:00:51	☆☆☆☆☆	
2024-04-04 14:26:27	Default	"<0014169645574>	1747	6133291747	Outgoing	00:00:05	☆☆☆☆☆	
2024-04-04 14:26:27	Default	"<0014169645574>	0014169645574	1747	Incoming	00:00:05	☆☆☆☆☆	
2024-04-03 14:39:30	Default	"<0014169645574>	1747	6133291747	Outgoing	00:00:05	☆☆☆☆☆	
2024-04-03 14:39:30	Default	"<0014169645574>	0014169645574	1747	Incoming	00:00:05	☆☆☆☆☆	
2024-04-03 14:38:22	Default	"<5132108100>	1747	6133291747	Outgoing	00:00:05	☆☆☆☆☆	
2024-04-03 14:38:22	Default	"<5132108100>	5132108100	1747	Incoming	00:00:05	☆☆☆☆☆	
2024-04-02 18:24:18	Default	"<21988250851>	21988250851	1216	Incoming	00:02:10	☆☆☆☆☆	
2024-04-02 18:24:18	Default	"<21988250851>	1216	6133291216	Outgoing	00:02:10	☆☆☆☆☆	
2024-04-02 17:51:42	Default	"<5132108100>	5132108100	1747	Incoming	00:00:07	☆☆☆☆☆	
2024-04-02 17:51:42	Default	"<5132108100>	1747	6133291747	Outgoing	00:00:07	☆☆☆☆☆	

Search

The calls displayed here are those of all the extensions that the user has the right to see.

Technical Specifications

Name	Attimo Recordings
Version	4.0.1-2
OS	Linux Debian 11 64 bits
Compatibility	Attimo CloudPBX
Database	MariaDB 10.x
Developed Language	PHP 8, HTML
Necessary Memory	2 GB
Necessary HDD	250 MB
Packet Technology	DEB
Dependency	Already installed in Attimo CloudPBX

Reports

Filters

In order to facilitate the search of recordings it is possible to create filters, which can be used at the time of the search.

The screenshot shows the 'Filters' configuration page in the Attimo CloudPBX interface. The left sidebar contains navigation links for 'admin', 'PANEL', 'REPORTS', and 'SETTINGS'. The 'REPORTS' link is highlighted. The main content area is titled 'Filters' and includes tabs for 'Panel' and 'Schedules Profiles'. The 'GENERAL' tab is active, showing fields for 'Description', 'Team' (set to 'All'), and 'Extensions'. Below these is an 'Items' section with a table for defining search criteria. The table has columns for 'Condition', 'Search By', 'Mode', and 'Value'. A row is currently defined with 'AND' as the condition, 'Called To' as the search field, and 'Begins With' as the mode. An 'Add' button is visible next to the row. A 'Save' button is located at the bottom right of the interface.

Condition	Search By	Mode	Value
AND	Called To	Begins With	

In Filters the options to take to configure are the following:

Description, brief description of the filter.

Extensions, Extensions to include in the filter.

Team, Team to include in the filter.

Items, Different search criteria.

Condition, AND or OR condition.

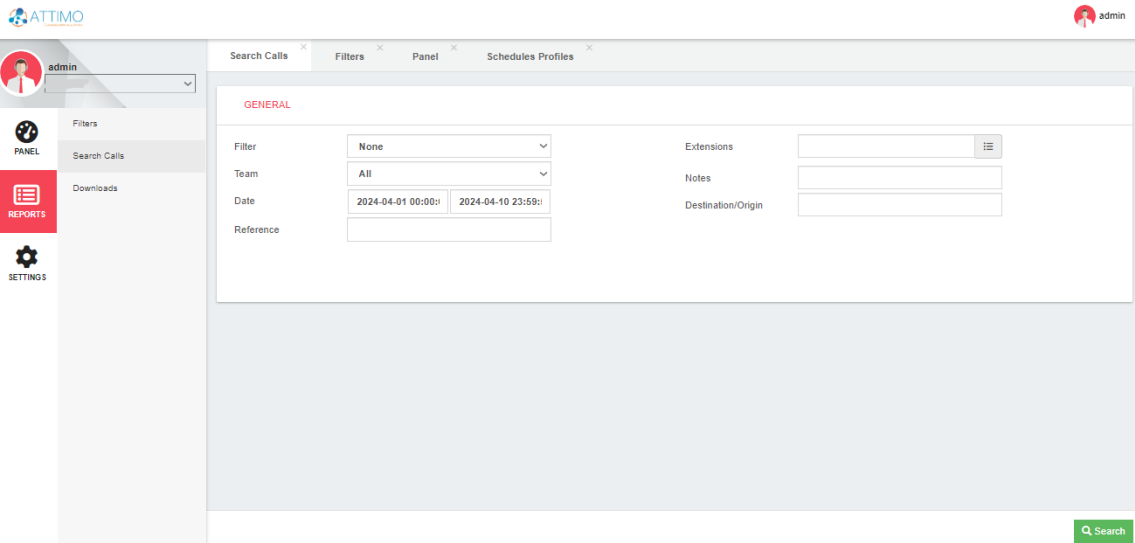
Search By, Search field.

Mode, Search Mode.

Value, value to search.

Search Calls

In Search Calls, reports are generated with the possibility to modify Notes, Flags, References, etc. These reports can be exported.



In Search Calls the options to take to configure are the following:

Filter, filter used to create the report. This filter was created previously.

Team, team to include in the search.

Date, date range to evaluate.

Reference, search for some reference in the recordings.

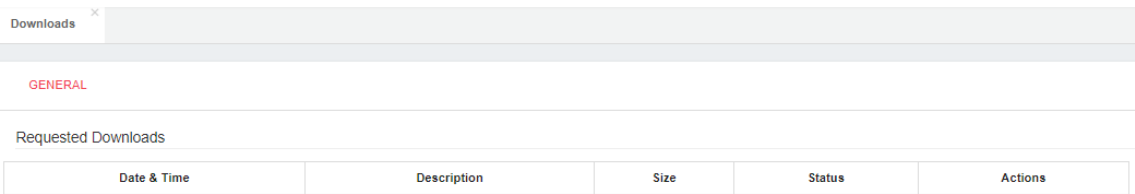
Extensions, extensions to include in the report.

Note, search for a note in the recordings.

Destination/Origin, called number or number where the call originated.

Downloads

When one selects the option to export a report, this is the area where exported reports appear, so these can be downloaded.



All exported reports contain the recordings and a PDF file with a list of these. It also includes the flags, notes, and references if they are available.

Settings

Flag Profiles

The Flag is used to categorize the recording. You can create multiple Flag profiles with different colors and flag meanings.

To create a Flag, go to Settings/Flag Profiles:

Flag Profiles

GENERAL

Description

Flags *

Name	Color
Sales	Black
Support	Blue
	Black

Add

In Flag Profiles the following data must be configured:

Description, brief description with which the Flag Profile is identified.

Add, press this button to add a Flag.

Name, flag name.

Color, Color with which this flag is associated.

Teams

It is possible to group extensions in Teams and each Team will have associated a Flag Profile. This greatly facilitates the administration of recordings by department.

To create a Team, go to Settings/Teams:

GENERAL

Description

Flag Profile

Default

In Teams the following data must be configured:

Description, Brief description with which the Team is identified.

Flag Profile, Flag Profile with which this Team is associated.

Synchronization

Now synchronize the extension, the demo just synchronizes 8 extensions, you can select the extension and associate it with your Team.

To synchronize the extension, go to Settings/Synchronization:

GENERAL

Team

Default

Extensions

In Synchronization the following data must be configured:

Team, Name of the Team that the extensions belong to.

Extensions, Select the extensions to assign to this Team.

Extensions

It is possible to change the configuration of an extension after synchronizing, for this you have to go to Settings/Extensions:

GENERAL

Extension Name

Teams

Default

Outgoing Recording

Yes

Incoming Recording

Yes

Internal Recording

Yes

In Extensions you can configure the following options:

Team, Name of the Team that the extensions belong to.

Outgoing Recording, If the outgoing calls are to be recorded.

Incoming Recording, If the incoming calls are to be recorded.

Internal Recording, If the internal calls are to be recorded.

User Profiles

Sonata Recordings allows us to create users with different types of privileges, to create the Users Profiles it is necessary to go to Settings/User Profiles.

GENERAL

Name

Type

Administrator

Permissions

Listening Recordings,Add Note...

Unselected Modules

Add all

Selected Modules

Remove all

Flag Profiles

×

Search Calls

×

Extensions

×

Users

×

Users Profiles

×

Teams

×

Panel

×

Downloads

×

Filters

×

Synchronization

×

Schedules Profiles

×

Name, name for this User Profile.

Type, it defines the type of profile. On the user’s dialog, depending on the selected user type, the profile dropdown will be populated with profiles that have the same type as the user type.

Agent, just for the extensions owner, with this profile just can manage your own call recordings.

Supervisor, the Supervisor profile can have access to the assigned Team.

Manager, the Manager profile can have access to the assigned Supervisor.

Super Administrator, the administrator of the system.

Permissions, it allows you to select what permissions or actions has the user.

Add Notes, allows the user to create notes related to the recording.

Add Flags, allows user to assign flags related to the recording.

Add References, allows the user to associate the recording with a reference.

Allow Listening Recording, allows the user to listen to recordings.

Create Reports, allows the user to create reports.

Allow Delete, allows user to delete recordings.

Selected Modules, select the modules that the user will be able to enter.

Select Permissions ✕

Add all

Remove all

Listening Recordings

✕

Add Notes

✕

Add Flags

✕

Add References

✕

Deleting Recordings

✕

Create Reports

✕

Cancel

Accept

Users

Sonata Recordings allows us to create users with different types of privileges, to create the Users it is necessary to go to Settings/Users. Each user will be associated with a User Profile, which must have been previously created.

GENERAL

SETTINGS

Full Name *

User Name *

Email

Password *

User Type

Administrator

▼

Profile

Administrator

▼

Startup Dialog

Flag Profiles

▼

Department

Tenant

FUNCEF

▼



Change

Remove

In Users the options to configure are the following:

- Full Name**, full name of user.
- User Name**, name used to login.
- Password**, your secure password for login.
- User Type**, we have 4 types of users:

Agent, just for the extension's owner, with this profile just can manage your own call recordings.

Supervisor, the Supervisor profile can have access to the assigned Team.

Manager, the Manager profile can have access to the assigned Supervisor.

Admin, the administrator of the system.

Profile, profile for this User.

Startup Dialog, which dialog to be displayed when logging into the system.

Department, user Department (Example: Development).

Tenant, tenant to whom belongs this user.

Extension, this option only shows if you select Agent. It is the extension associated with the Agent, an Agent can only view its recordings.

Teams, this option only shows if you select Supervisor. The Supervisor may have several Teams under his or her charge, which will give him access to the recordings that will have access to these Teams.

Supervisors, this option only shows if you select Manager. The Manager may be in charge of several supervisors, which will give access to the recordings that these supervisors have access to.

Select Image, user avatar.

First create the Agent, one for each extension. Remember to associate the user with each extension. In the second tab you can set the Language and more settings.

The screenshot shows a user configuration interface with two tabs: 'GENERAL' and 'SETTINGS'. The 'GENERAL' tab is active. Below the tabs, there are three settings:

- Language**: A dropdown menu showing 'English'.
- Timezone**: A dropdown menu showing '(GMT -3:00) America/Sao_Paulo'.
- Multitab**: A toggle switch labeled 'Yes'.

In second tab of Users the options to take to configure are the following:

Language, Language to display the interface in.

TimeZone, Time zone that the recordings will be displayed in.

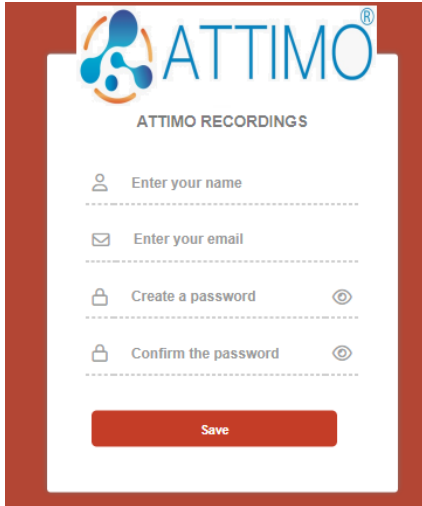
Multitab, the MultiTab option allows several forms to be displayed at the same time on the screen with the possibility of switching from one to the other by clicking the TAB.

Installation

In your Attimo CloudPBX go to Admin/Add-ons/Add-ons, select Attimo Recording and press the install button  shown in the **Actions** column. Wait a couple of minutes and then click the Sonata Recordings link and start configuring.

Press URL

Press the URL in the Add-ons list.

A registration form for Attimo Recordings. At the top is the Attimo logo and the text "ATTIMO RECORDINGS". Below this are four input fields: "Enter your name" (with a person icon), "Enter your email" (with an envelope icon), "Create a password" (with a lock icon and a toggle for visibility), and "Confirm the password" (with a lock icon and a toggle for visibility). At the bottom is a red "Save" button.

The first time we enter we must add the following information.

Enter your name, Enter your full name.

Enter your email, Enter your email address.

Create a password, Enter the password to be used by the admin user.

Confirm the password, Confirmation of password entered previously.

Login with the Username **admin** and password previously created.

A login form for Attimo Recordings. At the top is the Attimo logo and the text "ATTIMO" with the tagline "Communication as a Service". Below this is the heading "LOG IN". There are two input fields: "Username" and "Password" (with a toggle for visibility). At the bottom is a red "Sign In" button.

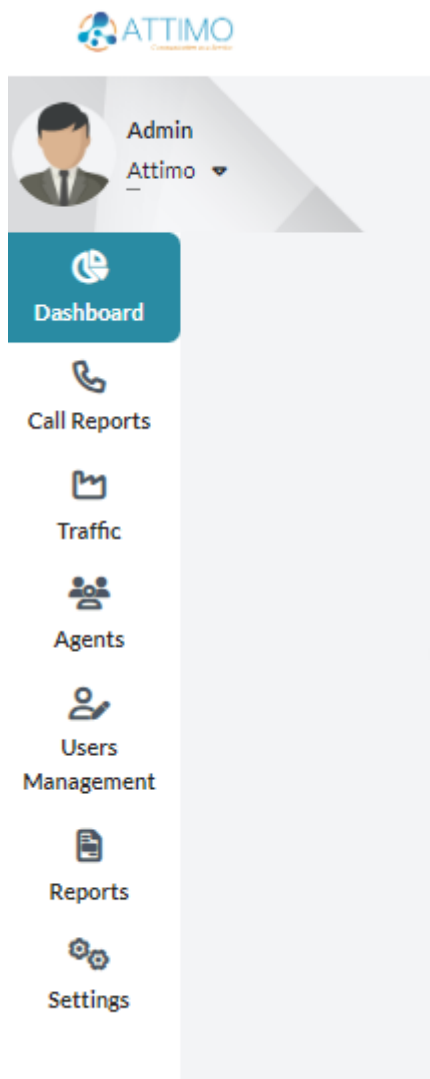


Stats

INDEX

Navigation Menu	3
Dashboard	6
Call Reports	7
Traffic	17
Agents	22
User Management	27
Reports	29
Settings	32

Navigation Menu



On the Navigation Menu, we got the following options.

Call Reports, Here, we can see all the calls taking into account the queues and agents:

Traffic, Here, we show reports for the call distribution considering the hours, days, days of the week, and month:

- **By Hour**, with this report we can evaluate the hours with the highest traffic on our Call Center. We can see a graph portraying this behavior.
- **By Day**, Shows the total number of calls by day in a graph form, and also provides information such as, answered calls, abandoned calls, average call time, among many others.
- **By Day Of Week**, Shows the total number of calls by the day of the week in graph form. As well as the answered calls, abandoned calls, average call time, among others.
- **By Month**, Shows the total number of calls per month in graph form, as well as the answered calls, abandoned calls, and average per call, among others.
- **Lost Calls By Hour**, Shows a summary of the lost calls by agent per hour.

Agents, Here, we can obtain varied information from our agents:

- **Session Details**, Session Detail per agent. Here, you can see how many times they logged into a queue, average session duration, pauses, average pause time, among others.
- **Availability**, Shows the availability of each agent. Here, you can see how many agents are evaluated, how much time they were in conversation, pause time, on hold, and available.
- **Pauses**, It shows detailed and summarized report of all the pauses of each agent in each queue.
- **Session By Hour**, Shows the total time in session by hour for each agent.
- **By Hour**, Here, we can see how many agents are available depending on the day and time. It also shows the list of agents that were available at those times.

User Management, Here, we create the system users with their respective privileges.

- **Roles**, Are privileges that will be assigned to the different users.
- **Users**, Here, we create the users with the role previously created.

Reports, Here, we configure some parameters used at the moment of generating the reports. Here, it is also possible to create automatic reports that are generated on specified date and time.

- **Shift**, It is possible to create specific work shifts, for example, if we have three (3) shifts on our Call Center, and let's say one of them is from 08:00 to 17:00, we can create that shift and apply it to any of the reports we saw previously.
- **Email Templates**, Since it is possible to create reports to be sent automatically by email, here, you would be able to create the templates for the email to send with each report.
- **Report Builder**, Here, we create the report that we wish to send through email automatically on a specified date and time.
- **Report Scheduler**, Here, we program the date, time, and repeat period to send the report built previously with the email template selected.

Settings, On this section we configure various settings:

- **Email Settings**, With this form you can configure the parameters needed for the email client to send the automatic reports.
- **Report Settings**, With this form you can configure the general parameters for the reports, such as the report headers, company logo, among others.
- **PBX Settings**, On this form you can configure the connection to the host from where we are going to obtain the information.

Summary, Shows the total of Answered Calls, Abandoned Calls, and the SLA (Service Level).

Service Level, Shows the Service Level in a graph form and more specific.

By Agent, Shows the detail of the calls by agent. How many calls were answered and not answered by each agent, average by call, and the duration of every call.

Lost Calls By Agent, Here, we show a list of the calls that were not answered by each agent and the different events for each call, for example, how much times did it spend ringing and if it was answered by another agent. It is important to notice that Lost Calls are not Abandoned Calls. Lost Calls did not necessarily leave the queue.

By Queue, Shows a list of all the queues and the total of attended calls, abandoned calls, hold times, and more. You can see the detail for each call in the queue.

Abandoned Calls, Shows the detail of all the abandoned calls and the causes for the abandon. For example, the caller hung up, hold time timed out, and many other forms that the caller left the queue.

Answered Calls Details, Shows a list of all the attended calls and the events the occurred for each call. These events are for example, the call enters the queue, it connects with an agent, the call is ended by the agent.

Outgoing Calls Details, Shows the detail of all the outgoing calls made by each agent.

Dashboard



0
Queues



0
Agents



0
Offered Calls

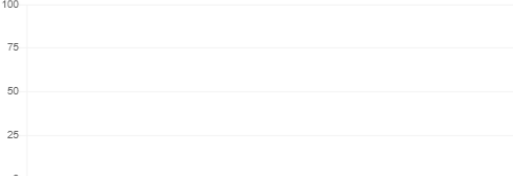


Answered Calls

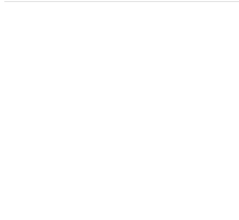


0
Abandoned Calls

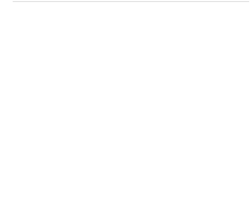
Service Level



Disconnection Causes



Abandoned Causes



Traffic By Hour



Lost Calls By Hour



Call Reports

Summary

Summarized report where in one screen we can see the total number of Answered and Abandoned Calls, as well as the SLA (Service Level)

Here, you can see the selected Queues, and the total of Answered and Abandoned calls, as well as the disconnection causes.

Call Reports Summary

Queues Considered	500501502503504505506507508509510511512513514515516517518520521522523524525526527
Total Calls Processed	2,407 answered / 173 abandoned
Percent	93.29% answered / 6.71% abandoned
Disconnection Causes	1,303 by Agent (54.13%) / 1,104 by Caller (45.87%)

We can also see the Service Level in 10 second intervals. These intervals can be configured under Settings > Report Settings.

Service Level

Call Type

All

Waiting Time (Seconds)	Number Calls	Delta	Percent
10	1960	0	75.97%
20	2199	+ 239	85.23%
30	2281	+ 82	88.41%
40	2338	+ 57	90.62%
50	2391	+ 53	92.67%
60	2427	+ 36	94.07%

Now, we see a summary of the Answered Calls, Average Call Time, Waiting Time, and additional information.

Answered Calls	
Answered calls	2407
Average calls length	00:02:29
Min calls length	00:00:00
Max calls length	00:21:01
Total calls length	99:54:17
Average calls waiting time	00:00:13
Min waiting time	00:00:03
Max waiting time	00:04:27
Total waiting time	08:41:59
Answered Calls	

We can also see a summary of the abandoned calls with the information of average waiting time and other additional data.

Abandoned Calls	
Abandoned calls	173
Average calls waiting time	00:00:46
Min waiting time	00:00:00
Max waiting time	00:03:05
Total waiting time	02:14:02
Abandoned calls	

Service level

In every Call Center, the Service Level (SLA) is one of the most important reports since a good SLA will guarantee us a good Customer Service. On this report it is possible to create filters to have specific parameters for the Service Level. The available options are the following:

More filters

×

SLA: initial period

2

SLA: max period

80

SLA: interval

2

Short Call Wait Limit

0

Short Call Talk Limit

0

Save

Cancel

SLA: initial period, from which second we wish to show the SLA, traditionally, it is shown from the 10th second.

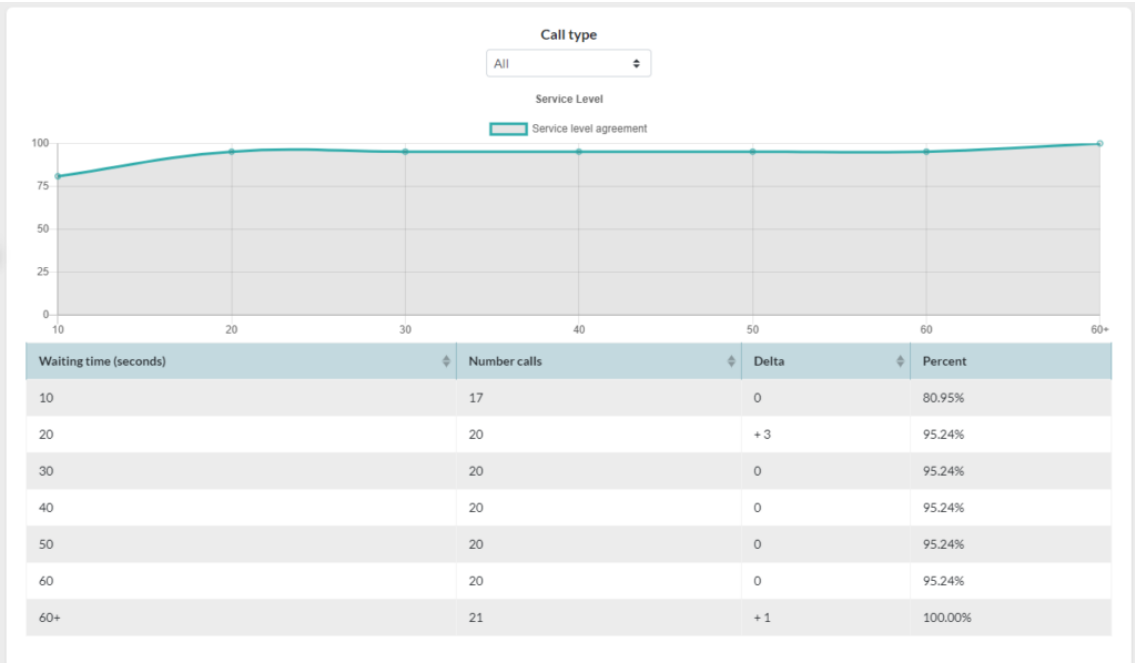
SLA: max period, Up to which number of seconds you wish to measure the SLA, by default this is 90 seconds.

SLA: interval, the interval to measure the SLA, meaning how often we wish to see information from the SLA. We recommend between 5 and 10 seconds.

Short Call Wait Limit, the minimum Call Wait time to consider a call as valid. Many times, the callers hang up the calls too fast and this could affect the statistics of our Call Center.

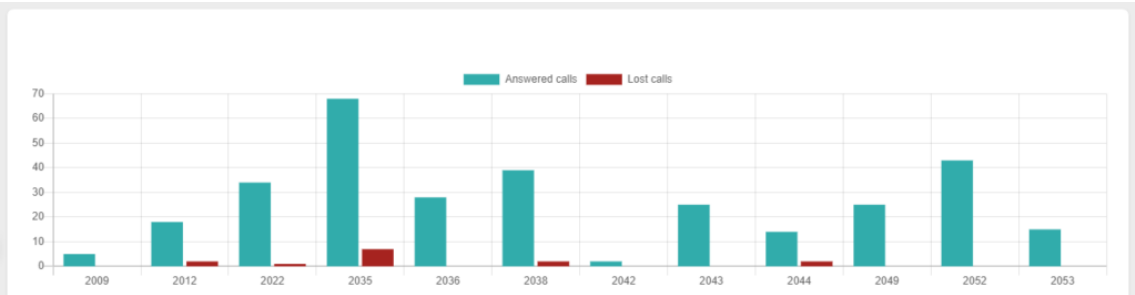
Short Call Talk Limit, the Minimum Call Time to consider the call as valid. On some occasions, the calls hang up the call as soon as the agent answers. This could affect the statistics of our Call Center.

Below, you can see a graph where you can observe the Service Level (SLA) behavior over time.



By Agents

These are summarized and detailed reports for the answered and lost calls by agent. Below, you can see a graph reflecting this information.



You can also see a detailed table with the following information:

- Answered Calls**, total number of calls answered by the agent.
- Answered Calls %**, call percentage of the answered calls with respect to all of the selected agents.
- Avg Call Time**, average duration of the answered calls.
- Lost Calls**, total number of calls not answered by an agent.
- Lost Calls %**, percentage of the calls not answered with respect to all the selected agents.
- Total Call Time**, duration of all the answered calls.

Agent	Answered calls	Answered calls %	Avg call time	Lost calls	Lost calls %	Total call time
+2009 Gregory Peck	64	2.66%	00:02:37	0	0.00%	02:47:42
+2012 Cate Blanchett	132	5.48%	00:02:37	4	2.70%	05:45:34
+2022 Spencer Tracy	269	11.18%	00:02:22	1	0.68%	10:38:33
+2027 Shah Rukh Khan	3	0.12%	00:01:03	1	0.68%	00:03:11
+2035 Viola Davis	376	15.62%	00:02:19	28	18.92%	14:32:09
+2036 Sophia Loren	225	9.35%	00:02:03	0	0.00%	07:44:36
+2038 Cary Grant	226	9.39%	00:02:48	72	48.65%	10:35:22
+2042 Marilyn Monroe	9	0.37%	00:03:20	0	0.00%	00:30:07
+2043 Laurence Olivier	220	9.14%	00:02:34	11	7.43%	09:26:58
+2044 James Stewart	134	5.57%	00:02:24	7	4.73%	05:23:00
+2049 Grace Kelly	147	6.11%	00:02:47	7	4.73%	06:51:19
+2052 Julie Andrews	369	15.33%	00:02:46	15	10.14%	17:03:42
+2053 Halle Berry	233	9.68%	00:02:11	2	1.35%	08:32:04
Total	2407	100.00%	00:31:51	148	100.01%	99:54:17

It is possible to see Call Details for the calls by each agent, by pressing the plus (+) symbol that is in front of the agent number.

2022 Spencer Tracy

34

10.76%

00:02:49

1

7.14%

01:36:14

PDF

Excel

CSV

Total Records: 6

All

Answered

Lost

Date	Queue	Caller	Hold time	Duration	Disconnection	Event	Position
2020-08-08 21:55:08	517 - INTERNATIONAL TECHNICAL	50325376640	00:00:05	00:00:30	Agent	COMPLETEAGENT	1
2020-08-10 07:22:09	501 - SUPER HEALTH	86889418	00:00:06	00:00:30	Caller	COMPLETECALLER	1
2020-08-10 10:22:03	518 - INTERNATIONAL HEALTH	50322187800	00:01:17	00:09:05	Caller	COMPLETECALLER	1
2020-08-10 15:45:41	501 - SUPER HEALTH	22556900	00:00:13	00:03:27	Agent	COMPLETEAGENT	1
2020-08-12 16:40:30	527 - FUNERAL BEER	89029250	00:00:07	00:01:45	Agent	COMPLETEAGENT	1
2020-08-14 17:26:18	501 - SUPER HEALTH	87969332	00:00:15	00:06:11	Caller	COMPLETECALLER	1

<

1

>

Lost Calls By Agents

On this report, we can see detailed information of calls not answered by an agent, with the following information:

- Date**, date and time for the call.
- Queue**, queue through which the call entered.
- Agent**, the agent that did not answer the call. Here, we got to notice that even if the agent did not answer the call, the call could have been answered by another agent. See the note after the image below.
- Caller**, number that called the Call Center.

148 lost calls			
Date	Queue	Agent	Caller
2020-08-08 04:35:13	505 - DANGEROUS AUTO	2038 - Cary Grant	86791806
2020-08-08 16:21:15	509 - BEER AUTO	2035 - Viola Davis	22558484
2020-08-08 16:57:12	510 - BEER HEALTH	2044 - James Stewart	22558484
2020-08-08 16:57:43	509 - BEER AUTO	2049 - Grace Kelly	22558461
2020-08-08 17:18:07	514 - SUPPLIERS	2035 - Viola Davis	88458845
2020-08-08 18:27:49	509 - BEER AUTO	2035 - Viola Davis	22558484
2020-08-08 18:35:35	514 - SUPPLIERS	2035 - Viola Davis	76625142
2020-08-08 18:35:56	514 - SUPPLIERS	2035 - Viola Davis	76625142
2020-08-08 19:23:09	509 - BEER AUTO	2044 - James Stewart	22558484
2020-08-08 19:51:54	500 - SUPER AUTO	2044 - James Stewart	84456834
1 2 3 4 ...			

If we observe well, each call has an icon on the last column. When you press this icon, we can investigate the whole trace of the call and where it ended. Given the possibility that in the end, it was answered by another agent or even the same agent. Below, you can see an example where agent 2044 did not answer the call, but the call was then answered by the agent 2036.

2020-08-08 16:57:12	510 - BEER HEALTH	2044 - James Stewart	22558484	
Track Caller in the rest of the day.				
Date	Agent	Event	Duration	
2020-08-08 16:57:12	-	ENTERQUEUE	00:00:00	
2020-08-08 16:57:12	2044 - James Stewart	RINGNOANSWER	00:00:00	
2020-08-08 16:57:36	2036 - Sophia Loren	CONNECT	00:00:24	
2020-08-08 16:58:23	2036 - Sophia Loren	COMPLETECALLER	00:00:47	

By Queue

On the reports by Queue, we can get a summary of all the activity in each queue with the following information:

Queue, Call Queue.

Answered Calls, total number of calls answered by the agents.

Abandoned Calls, total number of calls not tended by the agents and left the Queue.

Offered Calls, total number of calls that entered the Queue.

Max Callers, maximum position of a caller waiting in Queue, lower is better.

Answered %, percentage of answered calls with respect to the calls that entered Queue.

Abandoned %, percentage of calls not tended to with respect to the calls that entered Queue.

Avg Calls Length, average duration of answered calls.

Total Calls Length, total duration of answered calls.

Avg Calls Waiting Time, average wait time.

Total Waiting Time, sum of all wait times in a Queue.

Service Level, Service Level with respect to the value entered when creating the Queue in Attimo CloudPBX.

Queue	Answered calls	Abandoned calls	Offered calls	Max callers	Answered %	Abandoned %	Avg Call Length	Total Call Length	Avg Call Waiting Time	Total Waiting Time	SLA target	Service level
+514 SUPPLIERS	973	74	1047	1	92.93%	7.07%	00:02:01	32:43:58	00:00:14	04:20:38	30	86.53%
+500 SUPER AUTO	275	25	300	1	91.67%	8.33%	00:02:34	11:47:40	00:00:14	01:13:41	30	83.67%
+509 BEER AUTO	767	50	817	1	93.88%	6.12%	00:02:49	36:08:43	00:00:13	03:09:31	30	86.78%
+505 DANGEROUS AUTO	156	9	165	1	94.55%	5.45%	00:02:53	07:32:16	00:00:14	00:39:16	30	84.85%
+503 SUPER HOME	6	0	6	1	100.00%	0.00%	00:03:58	00:23:50	00:00:08	00:00:49	30	100.00%
+510 BEER HEALTH	123	8	131	1	93.89%	6.11%	00:02:59	06:08:46	00:00:21	00:47:27	30	77.86%
+517 INTERNATIONAL TECHNICAL	6	1	7	1	85.71%	14.29%	00:03:10	00:19:00	00:00:32	00:03:47	30	85.71%
+507 DANGEROUS HEALTH	66	5	71	1	92.96%	7.04%	00:02:49	03:06:22	00:00:27	00:32:26	30	78.87%
+501 SUPER HEALTH	19	0	19	1	100.00%	0.00%	00:02:07	00:40:24	00:00:12	00:03:52	30	89.47%
+526 VALUE PHARMACY	1	0	1	1	100.00%	0.00%	00:06:16	00:06:16	00:00:07	00:00:07	30	100.00%
+527 FUNERAL BEER	5	0	5	1	100.00%	0.00%	00:02:09	00:10:47	00:00:13	00:01:09	30	80.00%
+525 TRAVEL INSURANCE	2	0	2	1	100.00%	0.00%	00:09:40	00:19:21	00:00:06	00:00:12	30	100.00%
+508 DANGEROUS HOME	5	1	6	1	83.33%	16.67%	00:03:08	00:15:43	00:00:16	00:01:40	30	83.33%
+518 INTERNATIONAL HEALTH	2	0	2	1	100.00%	0.00%	00:05:31	00:11:02	00:00:41	00:01:22	30	50.00%
+508 THE OAK AUTO	1	0	1	1	100.00%	0.00%	00:00:09	00:00:09	00:00:04	00:00:04	30	100.00%
Total	2407	173	2580	10	93.29%	6.71%	00:02:13	99:54:17	00:04:02	10:56:01		85.80%

It is possible to see a call detail of the calls in each queue, by pressing the plus (+) symbol in front of the queue number.

Date	Queue	Hold time	Duration	Position	Disconnection	Event	Agent	Caller
2020-08-08 05:21:01	514 - SUPPLIERS	00:00:09	00:02:27	1	Caller	COMPLETECALLER	2038 - Cary Grant	82310703

Call events

Date	Duration	Event	Agent
2020-08-08 05:18:25	00:00:00	ENTERQUEUE	NONE
2020-08-08 05:18:34	00:00:09	CONNECT	2038 - Cary Grant
2020-08-08 05:21:01	00:02:27	COMPLETECALLER	2038 - Cary Grant

Abandoned Calls

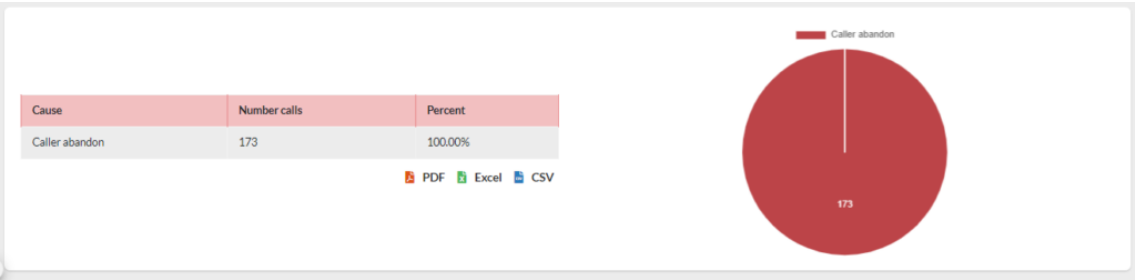
This is a detailed report for the abandoned calls on the Queues and their respective cause. Below, you can see a graph, we got to take into account that the possible causes for abandoned calls are the following:

Caller Abandoned, means that the caller did not want to continue waiting in Queue, and hung up.

Exit With Timeout, means that the caller waited the maximum timeout time configured on the Queue, and Queue sent the call to its final destination.

Exit With Key, means that the caller pressed a key to leave the Queue and take an alternative to their query. This happens when you configure an IVR with a message letting the caller know they can leave the Queue at any time to go elsewhere.

Exit With Empty, means that when the caller entered Queue there were no agents available and their call was rejected.



Next, we will see a table with the details of the abandoned calls with the following information:

- Date**, date and time of the abandoned call.
- Queue**, Queue where the call was abandoned.
- Caller**, caller’s phone number.
- Position**, caller’s position when they abandoned the call.
- Disconnection**, cause for abandon from the caller.
- Wait**, time the caller was waiting on Queue.

Sort: none -- Asc Per Page: 10

PDF Excel CSV

Abandoned Calls (173)

Date	Queue	Caller	Position	Disconnection	Wait
+ 2020-08-08 08:49:00	509 - BEER AUTO	22558484	1	Caller abandon	00:00:16
+ 2020-08-08 10:45:13	500 - SUPER AUTO	89441630	1	Caller abandon	00:00:10
+ 2020-08-08 11:10:07	507 - DANGEROUS HEALTH	81607866	1	Caller abandon	00:00:38
+ 2020-08-08 11:19:05	509 - BEER AUTO	22558484	1	Caller abandon	00:00:03
+ 2020-08-08 14:32:32	507 - DANGEROUS HEALTH	88858250	1	Caller abandon	00:01:19
+ 2020-08-08 15:42:40	509 - BEER AUTO	22558484	1	Caller abandon	00:01:01
+ 2020-08-08 16:06:21	500 - SUPER AUTO	83536963	1	Caller abandon	00:00:11
+ 2020-08-08 16:58:31	509 - BEER AUTO	22558461	1	Caller abandon	00:01:17
+ 2020-08-08 18:02:20	505 - DANGEROUS AUTO	81183988	1	Caller abandon	00:00:05
+ 2020-08-08 18:10:43	509 - BEER AUTO	22558484	2	Caller abandon	00:00:50

Sonata Stats - 2020

1 2 3 4 ...

It is possible to see the detail of the events for the abandoned calls by pressing the plus (+) symbol in front of the date.

Date	Queue	Caller	Position	Disconnection	Wait
+ 2020-08-08 08:49:00	509 - BEER AUTO	22558484	1	Caller abandon	00:00:16

Call Events

Date	Duration	Event	Agent
2020-08-08 08:48:44	00:00:00	ENTERQUEUE	NONE
2020-08-08 08:49:00	00:00:16	ABANDON	NONE

It is also possible to see if this call was answered later the same day by pressing the phone icon (☎) that is to the right of each call.

Abandoned Call Tracking

Agent	Date	Event	Duration	Queue
2035 - Viola Davis	2020-08-08 10:50:03	COMPLETEAGENT	00:04:15	500 - SUPER AUTO

+ 2020-08-08 11:10:07	507 - DANGEROUS HEALTH	81607866	1	Caller abandon	00:00:38
+ 2020-08-08 11:19:05	509 - BEER AUTO	22558484	1	Caller abandon	00:00:03
+ 2020-08-08 14:32:32	507 - DANGEROUS HEALTH	88858250	1	Caller abandon	00:01:19
+ 2020-08-08 15:42:40	509 - BEER AUTO	22558484	1	Caller abandon	00:01:01
+ 2020-08-08 16:06:21	500 - SUPER AUTO	83536963	1	Caller abandon	00:00:11
+ 2020-08-08 16:58:31	509 - BEER AUTO	22558461	1	Caller abandon	00:01:17
+ 2020-08-08 18:02:20	505 - DANGEROUS AUTO	81183988	1	Caller abandon	00:00:05
+ 2020-08-08 18:10:43	509 - BEER AUTO	22558484	2	Caller abandon	00:00:50

Answered Calls Details

This is a detailed report of the calls entering each Queue, with the possibility of seeing all of the events related to the call with the following information:

- Date**, date and time of the call.
- Queue**, queue through which the call entered.
- Caller**, caller’s phone number.
- Position**, caller’s position when the call was tended to.
- Disconnected By**, who released the call, the agent or the caller.
- Wait**, wait time before the call was tended to.
- Duration**, total call duration.
- Agent**, agent that tended the call.

Sort -- none -- Asc Per Page 10

Answered Calls (2407)

Date	Queue	Caller	Position	Disconnected by	Wait	Duration	Agent
+ 2020-08-08 04:40:39	505 - DANGEROUS AUTO	86791806	1	Caller	00:00:37	00:05:04	2038 - Cary Grant
+ 2020-08-08 04:47:11	500 - SUPER AUTO	86791806	1	Caller	00:00:09	00:02:39	2038 - Cary Grant
+ 2020-08-08 05:21:01	514 - SUPPLIERS	82310703	1	Caller	00:00:09	00:02:27	2038 - Cary Grant
+ 2020-08-08 05:54:55	514 - SUPPLIERS	82310703	1	Caller	00:00:08	00:05:10	2038 - Cary Grant
+ 2020-08-08 06:08:26	514 - SUPPLIERS	85352513	1	Agent	00:00:09	00:02:52	2038 - Cary Grant
+ 2020-08-08 06:15:03	514 - SUPPLIERS	88647514	1	Caller	00:00:07	00:02:59	2038 - Cary Grant
+ 2020-08-08 07:06:39	500 - SUPER AUTO	89454141	1	Agent	00:00:11	00:03:57	2035 - Viola Davis
+ 2020-08-08 07:08:57	509 - BEER AUTO	22558484	1	Caller	00:00:07	00:06:07	2049 - Grace Kelly
+ 2020-08-08 07:23:03	509 - BEER AUTO	22558484	1	Caller	00:00:06	00:01:37	2049 - Grace Kelly
+ 2020-08-08 07:39:32	509 - BEER AUTO	22558484	1	Agent	00:00:08	00:02:03	2035 - Viola Davis

Sonata Stats - 2020

<

<

1

2

3

4

...

>

>

It is possible to see a detail of the events for each call by pressing the Plus (+) symbol located in front of the date.

Date	Queue	Caller	Position	Disconnected by	Wait	Duration	Agent
+ 2020-08-08 04:40:39	505 - DANGEROUS AUTO	86791806	1	Caller	00:00:37	00:05:04	2038 - Cary Grant

Call events

Date	Duration	Event	Agent
2020-08-08 04:34:58	00:00:00	ENTERQUEUE	NONE
2020-08-08 04:35:13	00:00:15	RINGNOANSWER	2038 - Cary Grant
2020-08-08 04:35:35	00:00:37	CONNECT	2038 - Cary Grant
2020-08-08 04:40:39	00:05:04	COMPLETECALLER	2038 - Cary Grant

Outgoing Calls Details

Here, we can see a detailed report from all of the outgoing calls by the agents. The information we can see is the following:

Date, date and time of the call.

Agent, agent that made the call.

Destination, number dialed by the agent.

Disposition, type of disposition for the call.

Duration, call duration.

Wait Time, ring time before the call was answered.

Trunk, trunk through which the call was made.

Outgoing Calls (3675)

Sort: -- none --

Asc

Per Page: 10

Date	Agent	Extension	Destination	Disposition	Duration	Wait time	Trunk
2020-08-07 18:04:02	2043 - Laurence Oliver	2043	084315482	ANSWERED	00:00:36	00:00:17	4
2020-08-07 18:04:59	2043 - Laurence Oliver	2043	085552678	ANSWERED	00:00:39	00:00:13	3
2020-08-07 18:07:15	2043 - Laurence Oliver	2043	084315482	ANSWERED	00:00:49	00:00:16	4
2020-08-07 18:13:07	2043 - Laurence Oliver	2043	089881362	ANSWERED	00:00:05	00:00:05	3
2020-08-07 18:13:19	2043 - Laurence Oliver	2043	089881362	ANSWERED	00:00:07	00:00:05	3
2020-08-07 18:13:32	2043 - Laurence Oliver	2043	089881362	ANSWERED	00:00:14	00:00:32	3
2020-08-07 18:14:20	2043 - Laurence Oliver	2043	089881362	ANSWERED	00:00:11	00:00:28	3
2020-08-07 18:15:05	2043 - Laurence Oliver	2043	077325721	ANSWERED	00:00:15	00:00:16	3
2020-08-07 18:21:12	2035 - Viola Davis	2035	018004040	ANSWERED	00:00:14	00:00:00	1
2020-08-07 18:26:45	2009 - Gregory Peck	2009	086760256	ANSWERED	00:02:44	00:00:15	3

Sonata Stats - 2020

1

2

3

4

...

Traffic

By Hour

On this report, we can observe in graph form the hours with peak time of traffic on the Call Center. This information is very useful, since it helps us take decisions to know how many agents we need depending on the time. It is also possible to divide this report in half hour increments.



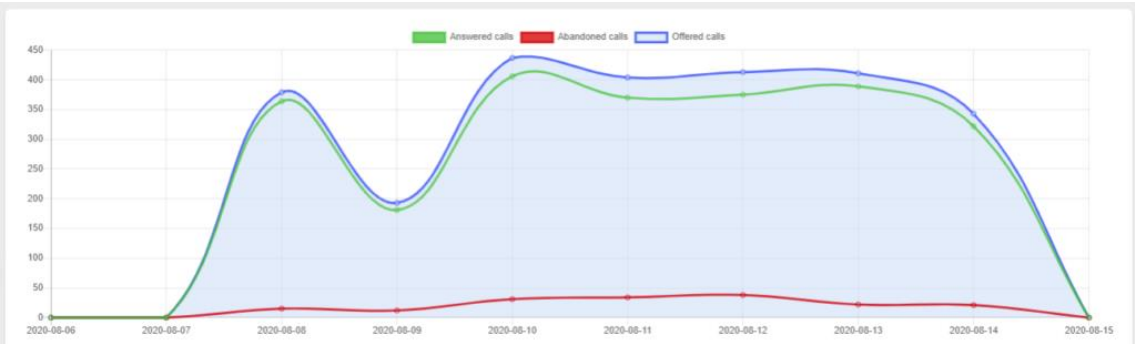
We can also see a table below the graph with the following information:

- Hour**, Hour to evaluate the information.
- Offered Calls**, number of calls that entered Queue.
- Answered Calls**, number of answered calls.
- Abandoned Calls**, number of abandoned calls.
- Answered Calls %**, percentage of calls answered on this time frame
- Avg Call Call Time**, average call duration by call.

Hour	Offered calls	Answered calls	Abandoned calls	Answered calls %	Avg call time	Min call time	Max call time
00:00	12	12	0	100.00%	00:03:23	00:00:31	00:08:16
01:00	11	11	0	100.00%	00:02:02	00:00:17	00:05:22
02:00	4	4	0	100.00%	00:03:34	00:01:13	00:05:18
03:00	9	8	1	88.89%	00:01:11	00:00:11	00:02:28
04:00	5	5	0	100.00%	00:02:46	00:00:44	00:05:04
05:00	25	21	4	84.00%	00:02:21	00:00:14	00:06:37
06:00	34	33	1	97.06%	00:02:16	00:00:02	00:05:08
07:00	114	106	8	92.98%	00:02:15	00:00:01	00:06:07
08:00	207	193	14	93.24%	00:02:22	00:00:02	00:09:45
09:00	202	192	10	95.05%	00:02:40	00:00:00	00:17:46
10:00	180	172	8	95.56%	00:02:36	00:00:01	00:18:26

Day

On this report we can get the calls offered (calls that entered the Queue), Answered Calls, and Abandoned calls totaled for the day.



We can also see a table below the graph with the following information:

- Day**, Date to show the information from.
- Offered Calls**, total number of calls that entered Queue.
- Answered Calls**, total number of calls answered by the agents.
- Abandoned Calls**, total number of calls not tended to, and left the Queue.
- Answered Calls %**, percentage of calls answered with respect to the calls that entered Queue.
- Avg Call Time**, average call time of answered calls.
- Min Call Time**, minimum duration of the answered calls.
- Max Call Time**, maximum duration of the answered calls.
- Avg Call Wait Time**, average Call Wait Time for all the calls.
- Min Call Wait Time**, minimum Call Wait Time for all the calls.
- Max Call Wait Time**, maximum Call Wait Time for all the calls.
- Abandoned Calls %**, percentage of the abandoned calls.
- Avg Abandoned Call Wait Time**, average Call Wait Time duration of abandoned calls.
- Min Abandoned Call Wait Time**, minimum Call Wait Time when a call is abandoned.
- Max Abandoned Call Wait Time**, maximum Call Wait Time when a call is abandoned.

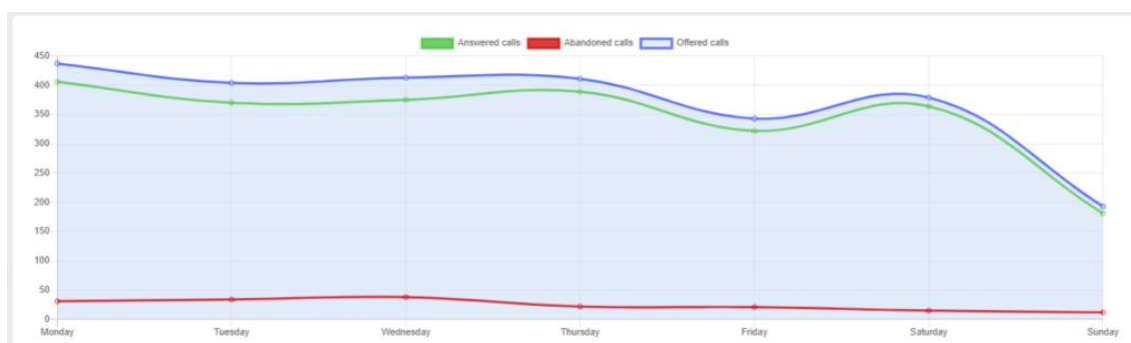
Day	Offered calls	Answered calls	Abandoned calls	Answered calls %	Avg call time	Min call time	Max call time	Avg call wait time	Min call wait time
2020-08-06	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
2020-08-07	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
2020-08-08	379	364	15	96.04%	00:02:27	00:00:00	00:12:10	00:00:12	00:00:04
2020-08-09	193	181	12	93.78%	00:02:09	00:00:01	00:08:16	00:00:10	00:00:04
2020-08-10	437	406	31	92.91%	00:02:39	00:00:00	00:17:46	00:00:15	00:00:04
2020-08-11	404	370	34	91.58%	00:02:37	00:00:02	00:13:31	00:00:15	00:00:04
2020-08-12	413	375	38	90.80%	00:02:35	00:00:02	00:21:01	00:00:13	00:00:04
2020-08-13	411	389	22	94.65%	00:02:32	00:00:00	00:13:48	00:00:10	00:00:04

On every table where there is a lot of information to show, you will find a pair of arrows indicating there is more information to either side.

Day of Week

On this report, we can get the offered calls (calls that entered the Queue), Answered Calls, and Abandoned Calls totaled by the day of the week.

This information is very important, since it helps us observe which day of the week there is more or less traffic, and so assign more or less agents.



Below the graph we can also see a table with information summarized by the day of the week with the following information:

Day, day of the week to show information from.

Offered Calls, total number of calls that entered Queue.

Answered Calls, total number of calls answered by the agents.

Abandoned Calls, total number of calls not tended to, and left the Queue.

Answered Calls %, percentage of calls answered with respect to the calls that entered Queue.

Avg Call Time, average call time of answered calls.

Min Call Time, minimum duration of the answered calls.

Max Call Time, maximum duration of the answered calls.

Avg Call Wait Time, average Call Wait Time for all the calls.

Min Call Wait Time, minimum Call Wait Time for all the calls.

Max Call Wait Time, maximum Call Wait Time for all the calls.

Abandoned Calls %, percentage of the abandoned calls.

Avg Abandoned Call Wait Time, average Call Wait Time duration of abandoned calls.

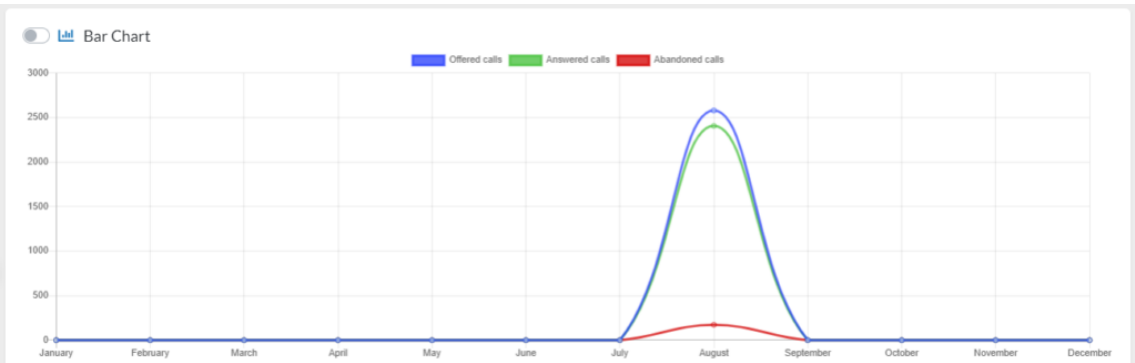
Min Abandoned Call Wait Time, minimum Call Wait Time when a call is abandoned.

Max Abandoned Call Wait Time, maximum Call Wait Time when a call is abandoned.

Day	Offered calls	Answered calls	Abandoned calls	Answered calls %	Avg call time	Min call time	Max call time	Avg call wait time	Min call wait time
Monday	437	406	31	92.91%	00:02:39	00:00:00	00:17:46	00:00:15	00:00:04
Tuesday	404	370	34	91.58%	00:02:37	00:00:02	00:13:31	00:00:15	00:00:04
Wednesday	413	375	38	90.80%	00:02:35	00:00:02	00:21:01	00:00:13	00:00:04
Thursday	411	389	22	94.65%	00:02:32	00:00:00	00:13:48	00:00:10	00:00:04
Friday	343	322	21	93.88%	00:02:10	00:00:02	00:08:41	00:00:11	00:00:03
Saturday	379	364	15	96.04%	00:02:27	00:00:00	00:12:10	00:00:12	00:00:04
Sunday	193	181	12	93.78%	00:02:09	00:00:01	00:08:16	00:00:10	00:00:04

By Month

On this report, you will get the offered calls (calls that entered the Queue), Answered Calls, and Abandoned Calls totaled by month.



We will also see a table with the information summarized by day of the week with the following information:

- Month**, month to show the information from.
- Offered Calls**, total number of calls that entered Queue.
- Answered Calls**, total number of calls answered by the agents.
- Abandoned Calls**, total number of calls not tended to, and left the Queue.
- Answered Calls %**, percentage of calls answered with respect to the calls that entered Queue.
- Avg Call Time**, average call time of answered calls.
- Min Call Time**, minimum duration of the answered calls.
- Max Call Time**, maximum duration of the answered calls.
- Avg Call Wait Time**, average Call Wait Time for all the calls.
- Min Call Wait Time**, minimum Call Wait Time for all the calls.
- Max Call Wait Time**, maximum Call Wait Time for all the calls.
- Abandoned Calls %**, percentage of the abandoned calls.
- Avg Abandoned Call Wait Time**, average Call Wait Time duration of abandoned calls.

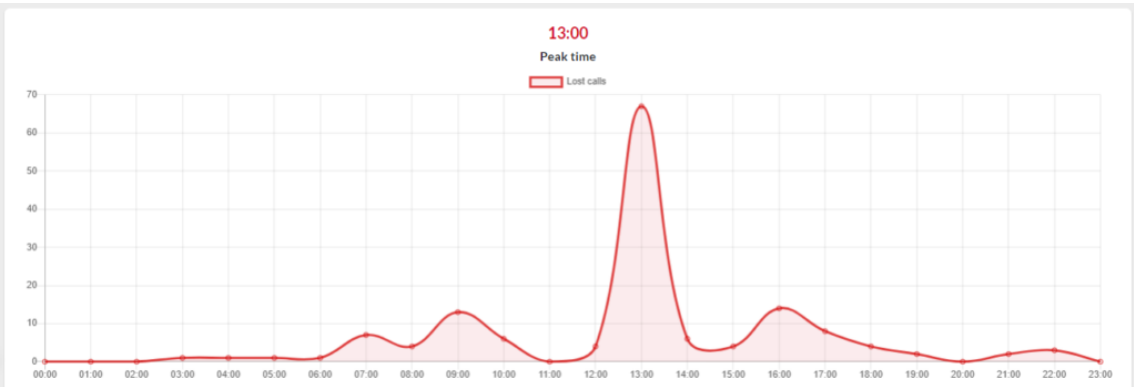
Min Abandoned Call Wait Time, minimum Call Wait Time when a call is abandoned.

Max Abandoned Call Wait Time, maximum Call Wait Time when a call is abandoned.

Month	Offered calls	Answered calls	Abandoned calls	Answered calls %	Avg call time	Min call time	Max call time	Avg call wait time	Min call wait time
January	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
February	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
March	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
April	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
May	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
June	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
July	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
August	2580	2407	173	93.29%	00:02:29	00:00:00	00:21:01	00:00:13	00:00:03
September	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
October	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
November	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00
December	0	0	0	0.00%	00:00:00	00:00:00	00:00:00	00:00:00	00:00:00

Lost Call By Hour

On this report we can observe the total number of calls lost every hour or every 30 minutes. This report is very important to determine if we need to have more agents on a determined time.



Agents

Session Detail

On this report we can see a summary, with the ability to see details for the agent’s sessions.

Report Details

Agent	Avg session time	Min session time	Max session time	Avg pause time
+ 2009 - Gregory Peck	0d:8h:52m:19s	0d:2h:24m:17s	0d:11h:56m:41s	0d:0h:6m:29s
+ 2012 - Cate Blanchett	0d:8h:11m:55s	0d:2h:54m:18s	0d:11h:59m:9s	0d:0h:6m:28s
+ 2022 - Spencer Tracy	0d:19h:35m:59s	0d:0h:1m:43s	6d:11h:26m:58s	0d:0h:10m:36s
+ 2027 - Shah Rukh Khan	0d:0h:4m:37s	0d:0h:0m:29s	0d:0h:8m:46s	0d:0h:2m:7s
+ 2035 - Viola Davis	0d:3h:48m:21s	0d:0h:5m:0s	0d:11h:36m:48s	0d:0h:8m:18s
+ 2036 - Sophia Loren	0d:6h:21m:36s	0d:0h:5m:14s	0d:11h:55m:5s	0d:0h:6m:16s
+ 2038 - Cary Grant	0d:17h:25m:18s	0d:0h:10m:45s	6d:13h:17m:24s	0d:0h:10m:19s
+ 2042 - Marilyn Monroe	0d:0h:19m:36s	0d:0h:8m:1s	0d:0h:42m:46s	0d:0h:5m:36s
+ 2043 - Laurence Olivier	0d:6h:57m:17s	0d:0h:1m:58s	0d:12h:0m:18s	0d:0h:7m:39s
+ 2044 - James Stewart	0d:9h:4m:34s	0d:5h:57m:27s	0d:14h:48m:51s	0d:0h:11m:24s
+ 2048 - Judy Garland	0d:0h:26m:36s	0d:0h:0m:41s	0d:1h:23m:19s	0d:0h:0m:0s
+ 2049 - Grace Kelly	0d:6h:20m:26s	0d:1h:0m:38s	0d:10h:52m:57s	0d:0h:14m:31s
+ 2052 - Jule Andrews	0d:10h:1m:33s	0d:4h:12m:0s	0d:12h:3m:30s	0d:0h:5m:58s
+ 2053 - Halle Berry	0d:12h:4m:43s	0d:12h:2m:10s	0d:12h:11m:9s	0d:0h:5m:25s

It is possible to see a detail of the events from each agent by clicking on the plus symbol (+) that is located in front of the agent number.

Agent	Avg session time	Min session time	Max session time	Avg pause time
2009 - Gregory Peck	0d:8h:52m:19s	0d:2h:24m:17s	0d:11h:56m:41s	0d:0h:6m:29s

Queue	Sessions	Sessions time	Avg session	Pauses	Pauses time	AVG pauses	Pause %
500 - SUPER AUTO	4	1d:11h:29m:15s	0d:8h:52m:18s	10	0d:1h:4m:59s	0d:0h:6m:29s	3.05%
501 - SUPER HEALTH	4	1d:11h:29m:17s	0d:8h:52m:19s	10	0d:1h:4m:59s	0d:0h:6m:29s	3.05%
502 - SUPER HOME	4	1d:11h:29m:17s	0d:8h:52m:19s	10	0d:1h:4m:59s	0d:0h:6m:29s	3.05%
503 - SAFE AUTO	4	1d:11h:29m:17s	0d:8h:52m:19s	10	0d:1h:5m:0s	0d:0h:6m:30s	3.05%
504 - SAFE HEALTH	4	1d:11h:29m:17s	0d:8h:52m:19s	10	0d:1h:5m:0s	0d:0h:6m:30s	3.05%
505 - DANGEROUS AUTO	4	1d:11h:29m:17s	0d:8h:52m:19s	10	0d:1h:5m:0s	0d:0h:6m:30s	3.05%
506 - DANGEROUS HOME	4	1d:11h:29m:17s	0d:8h:52m:19s	10	0d:1h:5m:0s	0d:0h:6m:30s	3.05%
507 - DANGEROUS HEALTH	4	1d:11h:29m:17s	0d:8h:52m:19s	10	0d:1h:5m:1s	0d:0h:6m:30s	3.05%
508 - THE OAK AUTO	4	1d:11h:29m:17s	0d:8h:52m:19s	10	0d:1h:5m:1s	0d:0h:6m:30s	3.05%
509 - BEER AUTO	4	1d:11h:29m:17s	0d:8h:52m:19s	10	0d:1h:5m:1s	0d:0h:6m:30s	3.05%

- The Possible events related to the agents are:
- START SESSION**, starts session on the Queues.
 - END SESSION**, ends session on the Queues.
 - PAUSE**, changes status to paused. This event possesses information that will allow us to determine the pause reason.
 - UNPAUSE**, End of pause.

Availability

On this report we can see the information about the agent’s availability, as well as the different task times.

On the following table, we can see how many agents were tending calls on the selected time range, as well as the average duration of all of the calls by agent.

Summary	
Agent sessions	
Numbers of agents evaluated	14
Average session time	0d:7h:49m:38s
Max session time	0d:0h:0m:0s
Min session time	0d:0h:0m:0s

Up next, we show a table with the information about the time usage for each of the agents. The information you will see is the following.

Agent, agent’s name and number.

Incoming Call Holdtime, total hold time.

Incoming Call Talktime, total talk time.

On Pause, total pause time.

Outgoing Call Holdtime, total hold time (or total ring time) on outgoing calls.

Outgoing Call Talktime, total conversation time for outgoing calls.

Productivity, the time percentage that the agent was in conversation with respect to the time in session.

Idle Time, total agent session time without the total pause time and time for incoming calls.

Time, total time that the agent was logged in to the system. Session Time.

Time %, time percentage that the agent was registered with respect to the other selected agents.

Agent	AVG session time	AVG pause time	AVG incoming call talktime	Min incoming call talktime	Max incoming call talktime	AVG incoming call holdtime	Min incoming call holdtime	Max incoming call holdtime	AVG outgoing call talktime	Min outgoing call talktime	Max outgoing call talktime
+ 2038 - Cary Grant	0d:17h:25m:18s	0d:0h:10m:19s	0d:0h:2m:5s	0d:0h:0m:0s	0d:0h:17m:30s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:47s	0d:0h:0m:0s	0d:0h:11m:0s
+ 2035 - Viola Davis	0d:3h:48m:21s	0d:0h:8m:18s	0d:0h:2m:7s	0d:0h:0m:0s	0d:0h:12m:10s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:40s	0d:0h:0m:0s	0d:0h:10m:40s
+ 2022 - Spencer Tracy	0d:19h:35m:59s	0d:0h:10m:36s	0d:0h:2m:20s	0d:0h:0m:0s	0d:0h:21m:1s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:1m:4s	0d:0h:0m:0s	0d:0h:10m:19s
+ 2036 - Sophia Loren	0d:6h:21m:36s	0d:0h:6m:16s	0d:0h:2m:3s	0d:0h:0m:0s	0d:0h:17m:46s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:51s	0d:0h:0m:0s	0d:0h:15m:33s
+ 2044 - James Stewart	0d:9h:4m:34s	0d:0h:11m:24s	0d:0h:2m:15s	0d:0h:0m:0s	0d:0h:8m:12s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:1m:5s	0d:0h:0m:0s	0d:0h:8m:33s
+ 2049 - Grace Kelly	0d:6h:20m:26s	0d:0h:14m:31s	0d:0h:2m:38s	0d:0h:0m:0s	0d:0h:10m:20s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:49s	0d:0h:0m:0s	0d:0h:4m:59s
+ 2052 - Julie Andrews	0d:10h:1m:33s	0d:0h:5m:58s	0d:0h:2m:37s	0d:0h:0m:0s	0d:0h:18m:26s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:1m:3s	0d:0h:0m:0s	0d:0h:10m:7s
+ 2053 - Halle Berry	0d:12h:4m:43s	0d:0h:5m:25s	0d:0h:2m:10s	0d:0h:0m:0s	0d:0h:13m:31s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:39s	0d:0h:0m:0s	0d:0h:5m:19s
+ 2012 - Cate Blanchett	0d:8h:11m:55s	0d:0h:6m:28s	0d:0h:2m:29s	0d:0h:0m:0s	0d:0h:8m:12s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:48s	0d:0h:0m:0s	0d:0h:6m:36s
+ 2027 - Shah Rukh Khan	0d:0h:4m:37s	0d:0h:2m:7s	0d:0h:0m:47s	0d:0h:0m:0s	0d:0h:1m:44s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:29s	0d:0h:0m:29s	0d:0h:0m:29s
+ 2048 - Judy Garland	0d:0h:26m:36s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:46s	0d:0h:0m:0s	0d:0h:2m:44s
+ 2009 - Gregory Peck	0d:8h:52m:19s	0d:0h:6m:29s	0d:0h:2m:37s	0d:0h:0m:14s	0d:0h:13m:13s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:50s	0d:0h:0m:0s	0d:0h:7m:8s
+ 2043 - Laurence Olivier	0d:6h:57m:17s	0d:0h:7m:39s	0d:0h:2m:26s	0d:0h:0m:0s	0d:0h:12m:45s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:43s	0d:0h:0m:0s	0d:0h:11m:42s
+ 2042 - Marilyn Monroe	0d:0h:19m:36s	0d:0h:5m:36s	0d:0h:3m:0s	0d:0h:0m:0s	0d:0h:6m:59s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:1m:28s	0d:0h:0m:2s	0d:0h:4m:7s

Pauses

It shows the causes of pauses by agents in each queue, their duration in detail and a summary.

Summary of Pauses by Agents.

Agent	Avg pause time	Frequent cause	Actions
2038 - Cary Grant	0d:0h:10m:19s	WC	Show Queues (22)
2035 - Viola Davis	0d:0h:8m:18s	WC	Show Queues (22)
2022 - Spencer Tracy	0d:0h:10m:36s	Administrative	Show Queues (27)
2036 - Sophia Loren	0d:0h:6m:16s	WC	Show Queues (27)
2044 - James Stewart	0d:0h:11m:24s	WC	Show Queues (27)
2049 - Grace Kelly	0d:0h:14m:31s	WC	Show Queues (22)
2052 - Julie Andrews	0d:0h:5m:58s	Administrative	Show Queues (22)
2053 - Halle Berry	0d:0h:5m:25s	WC	Show Queues (27)
2012 - Cate Blanchett	0d:0h:6m:28s	WC	Show Queues (27)
2027 - Shah Rukh Khan	0d:0h:2m:7s	WC	Show Queues (2)
2009 - Gregory Peck	0d:0h:6m:29s	WC	Show Queues (27)
2043 - Laurence Olivier	0d:0h:7m:39s	WC	Show Queues (27)
2042 - Marilyn Monroe	0d:0h:5m:36s	Administrative	Show Queues (26)

1

Summary of pauses by Agents in each Queue.

2027 - Shah Rukh Khan		0d:0h:2m:7s	WC	Hide
Queues				
Queue	Frequent cause	Avg pause time	Total pause time	Pauses
514 - SUPPLIERS	WC	0d:0h:2m:7s	0d:0h:2m:7s	1 Show Reasons
524 - FLEXY-AUTOS	WC	0d:0h:2m:7s	0d:0h:2m:7s	1 Show Reasons
1				

Details of each pause per agent in each queue.

2027 - Shah Rukh Khan

0d:0h:2m:7s

WC

Hide

Queues

Queue	Frequent cause	Avg pause time	Total pause time	Pauses
514 - SUPPLIERS	WC	0d:0h:2m:7s	0d:0h:2m:7s	1 Hide

Reasons

Reason	Queue	Total pause time	Avg pause time	Pauses
WC	514 - SUPPLIERS	0d:0h:2m:7s	0d:0h:2m:7s	1 Hide

Details

Reason	Start time	End time	Duration
WC	2020-08-10 09:02:48	2020-08-10 09:04:55	0d:0h:2m:7s

<

<

1

>

>

524 - FLEXY-AUTOS

WC

0d:0h:2m:7s

0d:0h:2m:7s

1 Show Reasons

<

<

1

>

>

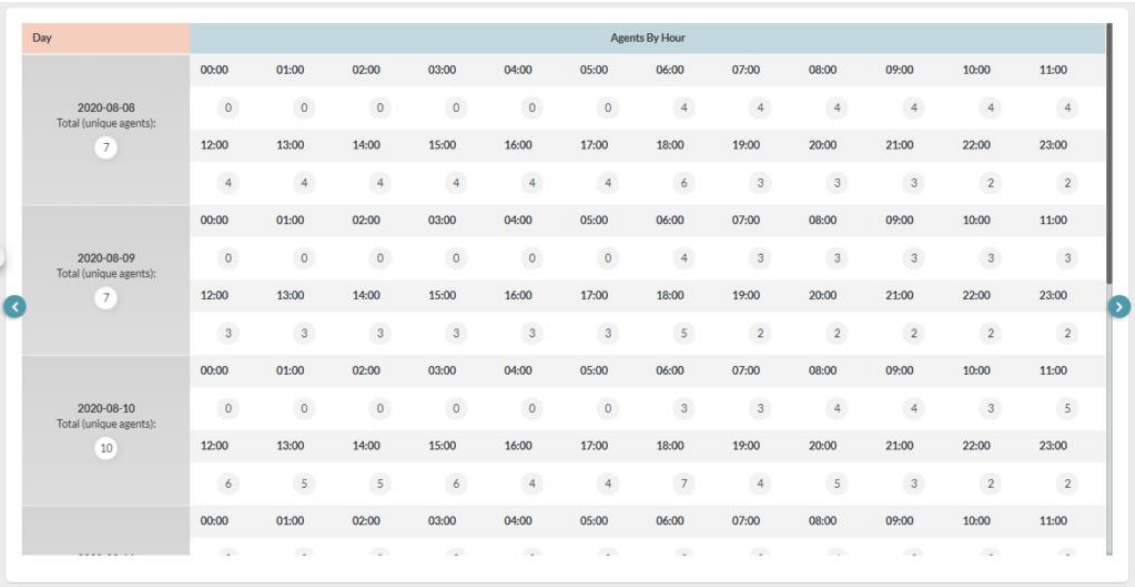
Session by Day

On this report we sum up the total time that an agent was logged in at a time.

Agent	Queue	Hours/Session Duration (HH:MM:SS)									
		00:00	01:00	02:00	03:00	04:00	05:00	06:00	07:00	08:00	09:00
2038 - Cary Grant Total: 1d:17h:33m:27s	SUPER AUTO - 500	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s	0d:0h:0m:0s
		12:00	13:00	14:00	15:00	16:00	17:00	18:00	19:00	20:00	21:00
		0d:2h:34m:10s	0d:5h:0m:0s	0d:5h:0m:0s	0d:5h:0m:0s	0d:5h:0m:0s	0d:4h:2m:14s	0d:4h:0m:0s	0d:3h:51m:0s	0d:3h:51m:0s	0d:3h:51m:0s
2038 - Cary Grant Total: 1d:17h:31m:44s	SUPER HOME - 502	00:00	01:00	02:00	03:00	04:00	05:00	06:00	07:00	08:00	09:00
		12:00	13:00	14:00	15:00	16:00	17:00	18:00	19:00	20:00	21:00
		0d:2h:34m:8s	0d:5h:0m:0s	0d:5h:0m:0s	0d:5h:0m:0s	0d:5h:0m:0s	0d:4h:0m:30s	0d:4h:0m:0s	0d:3h:51m:1s	0d:3h:51m:1s	0d:3h:51m:1s
2038 - Cary Grant Total: 1d:17h:31m:44s	SAFE AUTO - 503	00:00	01:00	02:00	03:00	04:00	05:00	06:00	07:00	08:00	09:00
		12:00	13:00	14:00	15:00	16:00	17:00	18:00	19:00	20:00	21:00
		0d:2h:34m:8s	0d:5h:0m:0s	0d:5h:0m:0s	0d:5h:0m:0s	0d:5h:0m:0s	0d:4h:0m:30s	0d:4h:0m:0s	0d:3h:51m:1s	0d:3h:51m:1s	0d:3h:51m:1s
2038 - Cary Grant Total: 1d:18h:23m:52s	DANGEROUS AUTO - 505	00:00	01:00	02:00	03:00	04:00	05:00	06:00	07:00	08:00	09:00
		12:00	13:00	14:00	15:00	16:00	17:00	18:00	19:00	20:00	21:00
		0d:2h:34m:8s	0d:5h:0m:0s	0d:5h:0m:0s	0d:5h:0m:0s	0d:5h:0m:0s	0d:4h:52m:38s	0d:4h:0m:0s	0d:3h:51m:1s	0d:3h:51m:1s	0d:3h:51m:1s

Session by Hour

On this report we can observe the total number of agents logged in to the system by day and time with the possibility to get a list of the agents on a determined day and time.



When pressing one of the circles we get the list of agents that were registered at that time and day.

Number of agents on 2020-08-08 at 12				
Agent	Queue	Start hour	End hour	Duration
2044 - James Stewart	SUPER HOME - 502	2020-08-08 06:43:02	2020-08-08 21:31:53	0d:14h:48m:51s
2049 - Grace Kelly	REMOTE AUTO - 512	2020-08-08 06:53:14	2020-08-08 13:53:54	0d:7h:0m:40s
2044 - James Stewart	DANGEROUS AUTO - 505	2020-08-08 06:43:02	2020-08-08 21:31:53	0d:14h:48m:51s
2036 - Sophia Loren	UNITY INSURANCE - 520	2020-08-08 06:33:23	2020-08-08 16:35:49	0d:10h:2m:26s
2036 - Sophia Loren	SUPER AUTO - 500	2020-08-08 06:33:23	2020-08-08 16:35:49	0d:10h:2m:26s
2035 - Viola Davis	EXTRA AUTO - 516	2020-08-08 07:08:54	2020-08-08 18:35:45	0d:11h:26m:51s
2035 - Viola Davis	ROBLE BREWERY - 515	2020-08-08 07:08:54	2020-08-08 18:35:45	0d:11h:26m:51s
2036 - Sophia Loren	EXTRA AUTO - 516	2020-08-08 06:33:23	2020-08-08 16:35:49	0d:10h:2m:26s
2049 - Grace Kelly	SUPER AUTO - 500	2020-08-08 06:53:13	2020-08-08 13:53:54	0d:7h:0m:41s
2036 - Sophia Loren	SAFE HEALTH - 504	2020-08-08 06:33:23	2020-08-08 16:35:49	0d:10h:2m:26s
2036 - Sophia Loren	BEER HEALTH - 510	2020-08-08 06:33:23	2020-08-08 16:35:49	0d:10h:2m:26s

When pressing the circle under the date it will give us the total number of agents that logged in that day.

Number of unique agents (6) on 2020-08-09		
Viola Davis 2035	Halle Berry 2053	Sophia Loren 2036
Spencer Tracy 2022	Elieth Elena Arcia 2052	Cate Blanchett 2012

User Management

Users

On this form we will proceed to create the system users.

Create User

Full Name

full name

Email

email

Username

username

Tenant

Password

password

Role

admin

Password Confirmation

password confirmation

System Permission

all

Save

The available options are as follows:

Full Name, User’s full name for descriptive purposes.

Email, User Email.

User Name, username required for the moment of login.

Tenant, the tenant with which this user is associated with.

Roles, the role created for this user.

Password and Password Confirmation, password used to log in to the system.

Roles

Roles are permissions that a user will have. Each user can have one or more roles associated with them.

Create Role

Role Name

role name

Permissions

Agents

Agent Availability

Agent Session Detail

Agent Session Time By Hour

Agents By Hour

Agent Pauses

Agent Session Time By Day

Call Reports

By Agent

By Queue

Lost Calls

Outgoing Calls

Service Level

Summary

Abandoned Calls

Answered Calls Details

Listen Outgoing Calls Recordings

Settings

Email Settings

Pbx Settings

Report Settings

Agent Synchronization

Reports

Email Templates

Report Builder

Report Scheduler

Shifts

License

License

Users Management

Roles

Users

System Permissions

Traffic

Lost Calls By Hour

Traffic By Day

Traffic By Day Of Week

Traffic By Hour

Traffic By Month

Save

The information to take into account when creating a role is:

Role Name, short name to identify the role.

Assign Permissions, permissions associated with this role.

Reports

Shifts

It is possible to create work shifts, for example, if our Call Center has three shifts and one of them is from 09:00 through 19:00, here we can create that shift so we can apply it to any of our reports.

Create Shift

Shift Name

shift name

Set shift days and hours

Time	AM											PM												
Day	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
Sunday																								
Monday																								
Tuesday																								
Wednesday																								
Thursday																								
Friday																								
Saturday																								

Drag to select hours

Reset selected

Save

To create a shift, we will need to configure the following options:

- Shift Name**, A name that will allow us to remember the shift.
- Set Shift Days And Hours**, Select by pressing on the grid the hours and days to include on the shift.

Email Templates

On this module, we can create the different templates to use when send the automatic emails. This template will be used when we configure the date and time to send the report by email on the Report Scheduler.

Create Template

Title

title

Subject

subject

Email Content

B I normal Sans Serif A = [icon] insert tag

write the content of your template here

this will be the content of the email, you can also add variables to the content.

Template preview

Save

The options that you can use to configure the Email Template are:

- Title**, The name to identify the template.
- Subject**, The subject used so that the receiver can identify the email received.
- Email Content**, Content for the email to send. Here, it is possible to add a number of variables. The available variables are:
 - user, it is substituted by the name of the sender.

to, it is substituted by the email address the email is sent to.

report_name, it is substituted by the name of the report sent.

repeat_mode, it is substituted by the repeat period.

startdate, it is substituted by the start date for the automatic emails.

link_to_report, it is substituted by the link to view the report online.

Report Builder

This module allows you to create profiles of different reports to send automatically.

 Create Report 

Name	name	Report	Summary	Call Type	All
SLA Initial Period (S)	10	SLA Interval (S)	10	SLA Max Period (S)	60
Minimum Call Time	0	Minimum Wait Time	0	Shift	select a shift
Queues	0 queues selected	Agents	0 agents selected	Format	pdf
Period	Yesterday				

 Save

The options you can configure so you build your report are the following:

Title, Title of the report. This title will be used to identify this report at the moment of creating the Report Schedule.

Report, type of report that we wish to send automatically.

Shift, Used if you wish to apply a shift to the report, you selected the desired shift.

Queues, Queues to include on the report.

Agents, Agents to include on the report.

Format, Format in which you want to send the report. This can be in PDF, Excel, or CSV.

Period, Date range for the report. Here, it is very important to take into consideration when selecting any option that is not 'custom,' this date will be validated by the date the report is being sent on. When using the custom option, the start date and end date will be used the first time the report is sent, however, if the scheduling on the **Report Scheduler** in **repeat mode** we choose weekly, these dates will have a week added, and so on depending on the **repeat mode** we select.

Report Scheduler

Here, we will schedule the date, time, and period through which a previously built report will be sent.

 Create 

Description	description	Report	report	Recipients	0 recipients
Start Date	Seg, 15 de Abr de 2024 14:0	Repeat Mode	daily	Every	1 day
Email Template	select an email template	Enabled	☒		

 send now  Save

These are the options you have to configure for the automatic reports:

Description, a brief description to identify the report schedule.

Report, report to be sent, previously created in **Report Builder**.

To, email to which we are going to send the report.

Start Date, date and time we wish to start sending the reports.

Repeat Mode, the interval in which we want to send the reports.

Every, how often we wish to apply the **Repeat Mode**, for example, if we wish to send it weekly, but being every two weeks, here we can input 2.

Weekdays, this option only appears if we have set **the Repeat Mode weekly**. Here, we select the days of the week we wish to perform the task.

Email Template, we select a previously created email template.

Enabled, Enable or disable the scheduled sending, This option is useful since it lets us stop the task without having to delete it completely.

Settings

Email Settings

On this module, you can configure the parameters for the email account where you will send the emails with the automatic reports.

Email Settings

Configuration Type

Sendmail

From Name

from name

From Email

from email

Bcc Email

bcc email

Send Email Test

email

Save

The information needed to configure the account is the following:

- From Name**, Name used to send the email.
- From Email**, Email address from the email is sent.

Bcc Email, If you desire to send a Blind Copy to another email address, you configure it here.

Send Email Test, you can send a test email to verify that the account has been configured correctly, you must add the email you wish to send the test to and then click on the envelope icon .

Report Settings

On this module you can configure the general system parameters, like headers, logos, and others.

Report Settings

Company Name

Attimo

Timezone

America/Sao Paulo

PDF Paper Size

A3

Week Starts On

monday

Memory Limit (MB)

128

Company logo



Remove

Change

Service Level Default Filters

Initial Period

10

Interval

10

Max Period

60

Call Wait Limit

0

Call Talk Limit

0

Save

The fields you can configure here are the following:

Company Name, The name for the company to be shown on the report header for the reports that are exported as PDF.

Timezone, The timezone to use when selecting the date and time for a report.

Week Starts On, refers to the day that is considered the beginning of the week. This is very useful for when selecting a date, particularly with options like This week or Last week.

Memory Limit (MB), this sets the maximum amount of memory in bytes that a script is allowed to allocate. This helps prevent poorly written scripts for eating up all available memory on a server. Note that to have no memory limit, set this directive to -1. Some processes require more memory resources (especially if you have a lot of data to process), increase memory if reporting delays.

We also have the ability to configure the defaults for the filters to apply on the reports, the options are:

Initial Period, Initial period to get reports for SLA (Service Level). This parameter indicates the first piece of information to get.

Interval, Time interval to the data for SLA (Service Level). Every one of this interval data shown.

Max Period, Maximum period to show data for SLA (Service Level). All of the data obtained after this value will be indicated with a 'value+', if the value is 80 for example, it will be shown as 80+.

PBX Settings

On this form, we configure the connection with the host where we are getting the information from.

PBX Settings

Host: localhost ✓ Port: 443 Secure: ☒

App Key: the key has been obtained from the server ✓

Required when the application is installed on a server separate from your PBX. ⓘ

Save

The option we have here are:

Host, if the Attimo Stats is installed on the same server where Attimo CloudPBX is installed, we recommend using 'localhost'.

Port, depending whether or not we are using HTTPS or not, we would choose the port we are going to use, usually 80 (HTTP) or 443 (HTTPS).

Secure, indicates whether you are using a secure connection (HTTPS) or not, usually when using this option, you are using port 443.

App Key, if Attimo Stats is on the same server as Attimo CloudPBX, it is not necessary to configure the APP key, since it is automatically generated. If Attimo Stats is installed on a server different to where Attimo CloudPBX is installed on, you would need to create the APP Key. You can do this from Attimo CloudPBX under Admin > Admin > Application Keys, create a new one, copy it and paste it on Attimo Stats.



Switchboard

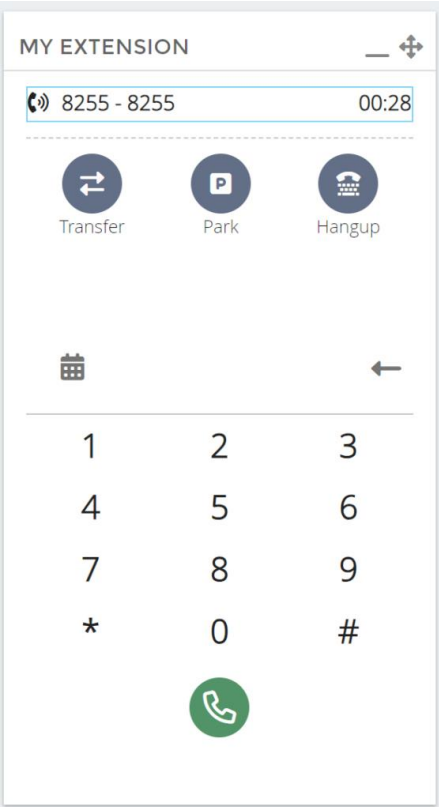
INDEX

Switchboard	3
Administration	12

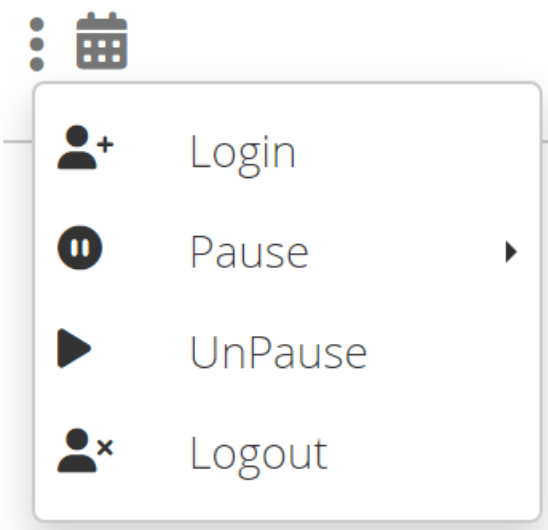
Switchboard

My Extensions

Shows my extension status and management; Transfer, Park and Hang-up (to activate these options you need to select the call).



If your extension belongs to a queue, the menu will appear to manage the Login, Logout, Pause and Unpause when you press the icon .



f you press the calendar  icon a call history will appear.

My Queues

Shows all the queues to which the extension belongs.

MY QUEUES				
Queue	Status	Calls Taken	Queued Calls	Pause Time
2902 - Music on Hold Test	LogOut Pause Pause With Reason		0	--:--:--

Possible actions (Right Click over the green, red, or yellow circle)

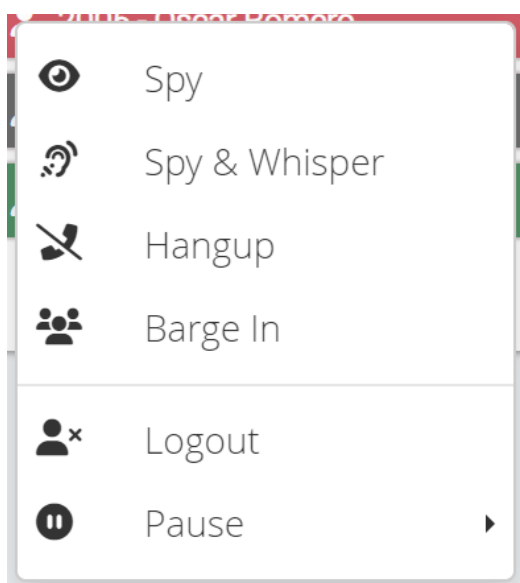
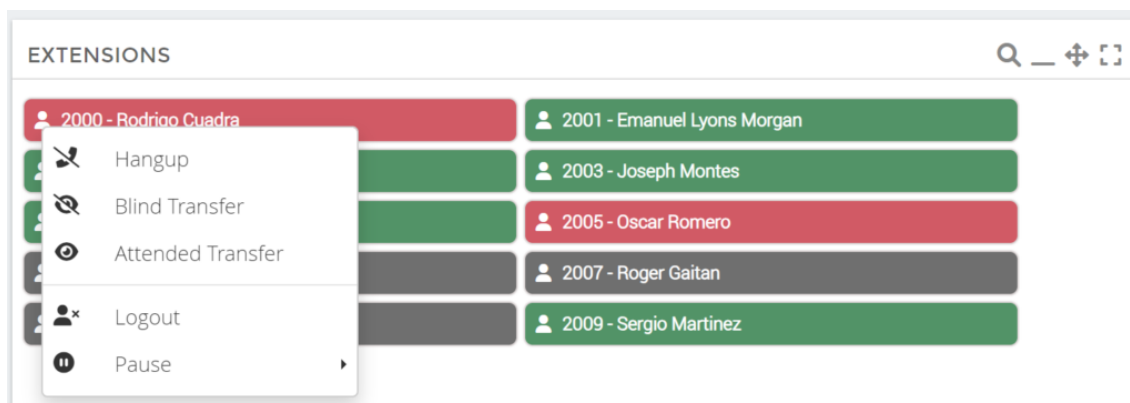
- Login**, Add the Agent to the Queue
- Logout**, Remove the Agent off the Queue
- Pause/UnPause**, Pause or UnPause the Agent on the Queue
- Pause with Reason**, Pause the Agent on the Queue with specify reason.

Extensions and Console Mode

It shows all extensions with their respective status with the possibility of interacting with them. You can hide extensions that are not registered in the system.

EXTENSIONS	
2000 - RODRIGO CUADRA 2005 - Oscar Romero 00:02 DND	2001 - EMANUEL LYONS MORGAN DND
2002 - ANTONIO MONTES DND	2003 - JOSEPH MONTES DND
2004 - MIGUEL RIVERA DND	2005 - OSCAR ROMERO 2000 - Rodrigo Cu... 00:01 DND
2006 - MAYNOR PERALTA DND	2007 - ROGER GAITAN DND
2008 - EMANUEL LYONS DELVALLE DND	2009 - SERGIO MARTINEZ DND

The Console mode only shows the extensions and their status, it does not show the number they are talking with, this is very useful to use as an operator console.



Possible actions (Right Click over the button)

Call, call to selected extension.

Blind Transfer, transfer the call to the selected extension.

Attended Transfer, transfer the call to the selected extension with supervision.

Spy, spy selected extension.

Spy & Whisper, Using Chanspy in “whisper” mode, the supervisor can speak directly to the agent without the client hearing.

Barge In, With this feature, the supervisor can join the call and speak to both the agent and the client.

Hang-Up, hang up the call.

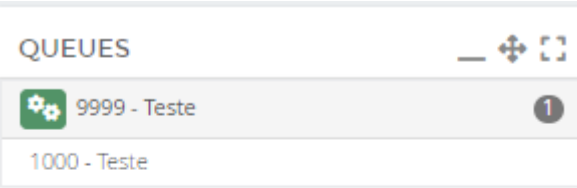
LogIn, Add the Agent to the Queue

Logout, Remove the Agent off the Queue

Pause/UnPause, Pause or UnPause the Agent on the Queue.

Queues

Show all Queues.



Possible actions (Right Click over Icon or Over the Member)

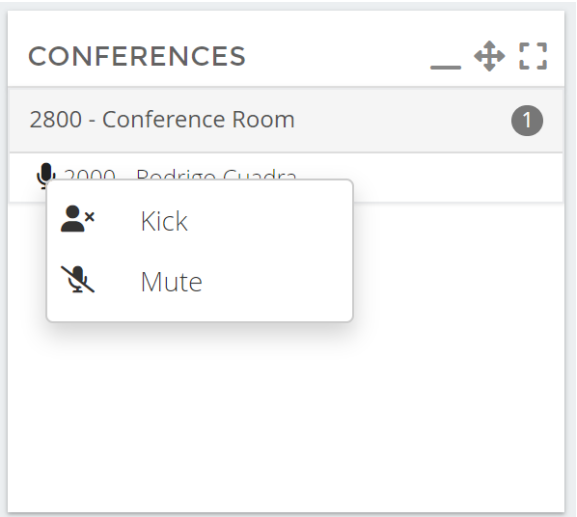
To add a Dynamic Member just click over the Icon of the Queue.

To Remove Dynamic Member just click over the Member and remove it.

To Pause or Un-Pause the Agent just click over the Agents.

Conferences

Show all the conferences.



Possible actions (Right Click over Conference Member)

Kick, remove the Member.

Mute, put the Member in mute.

Un-Mute, remove the mute from the member.

Parking Lots

Show all the Parking Lots

PARKING LOTS

700 - Default Parking

Possible actions (Right Click over parking Number)

Pickup, take the call.

Queues Stats Summary

Show complete information about the Queues.

QUEUES STATS SUMMARY										
Queue	Strategy	Loggedin Members	Available Members	Queued Calls	Completed Calls	Abandoned Calls	Service Level	Longest Hold Time	Hold Time	Talk Time
2900 - VitalPBX English	Round Robin Ordered	1	1	0	0	0	0.0	0	0	0
2901 - VitalPBX Spanish	Round Robin Ordered	2	2	0	0	0	0.0	0	0	0
2902 - Music on Hold Test	Round Robin Memory	1	1	0	0	0	0.0	0	0	0

Information:

Queue, number and name of the Queue.

Strategy, Ring Strategy of the Queue.

Logged In members, Quantity of members that are logged in the Queue.

Available Members, Quantity of members that are available in the Queue.

Queue Calls, current calls in the Queue.

Completed Calls, completed calls in the Queue.

Abandoned Calls, abandoned calls in the Queue.

Service Level, percentage of calls answered before a certain quantity of seconds.

Longest Hold Call, call that has the most time in queue.

Hold Time, total hold time.

Talk Time, total talk time.

Queue Members Summary

Show all the Agents with their current status.

QUEUE MEMBERS SUMMARY - 2900									
Member	Status	InCall	Calls Taken	Last Call	Paused Reason	Pause Time	Last Login	Calls History	
2003 - Joseph Montes		No	0	--		--:--:--	3 days ago		
2005 - Oscar Romero		--	0	--	--	--:--:--	--		

Information:

Member, extension and name of the Agent.

Status, Green indicates availability, Red indicates unavailability, and Yellow indicates a pause.

InCall, show if Agent is in call.

Calls Taken, quantity of calls taken by the Agent.

Last Call, show the last call attended by the Agent.

Pause Reason, if the Agent is in pause, it shows the reason.

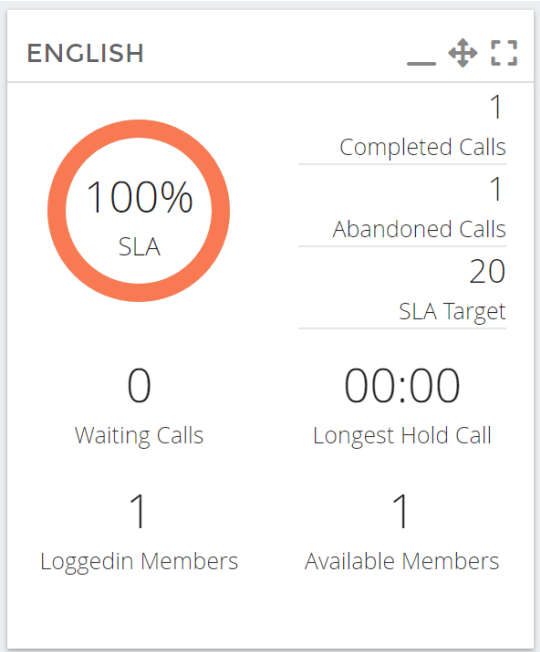
Pause Time, How long the Agent has been paused.

Last Login, How long has the Agent been registered in the Queue.

Call History, Shows a history of the Agent’s calls.

Queue Overview

Show statistics information about the Queue.



Information:

Service Level, percentage of call answered before certain quantity of seconds.

Completed Calls, completed calls in the Queue.

Abandoned Calls, abandoned calls in the Queue.

SLA Target, Number of seconds before which the call should be answered.

Waiting Calls, current calls in the Queue.

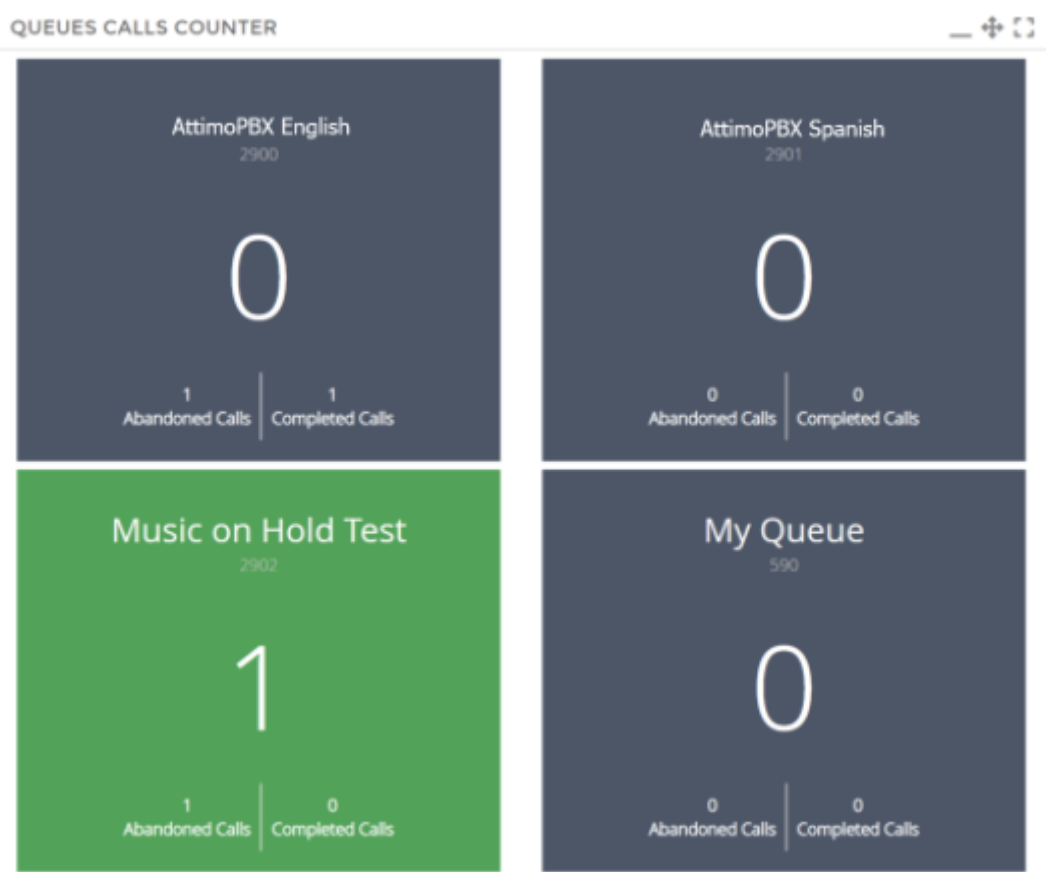
Longest Hold Call, call that has the most time waiting in queue.

Logged In Members, Quantity of members that are logged in the Queue.

Available Members, Quantity of members that are available in the Queue.

Queue Call Counter

Show real time Queue calls counter.



Information:

Big Number, current calls in the Queue.

Abandoned Calls, abandoned calls in the Queue.

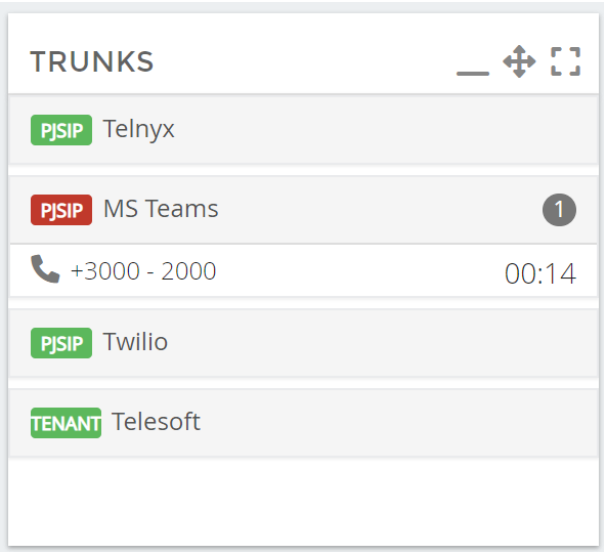
Completed Calls, completed calls in the Queue.

Audio an incoming call.

Trunk

In the trunk Widget you can see the activity of the PJSIP and IAX trunks. Below we show an image of how it would look when there is activity.

The green box shows the trunk technology, the gray circle shows the number of busy channels.



Queue Wallboard

Shows the status of each queue in real time. The data shown is shown below:



Extensions Summary

Shows a list of the extensions selected when creating the Widget. Details are shown below:

EXTENSIONS SUMMARY											
Extension	Reg. Status	DND	Q. Status	In Call	N. Out Calls	N. In Calls	Total Calls	Last Out Calls	Last In Calls	Pause Reason	Details
2000 - Rodrigo Cuadra				No	2	0	2	4 minutes ago	--	---	
2001 - Emanuel Lyons Morgan				No	0	0	0	--	--	--	
2002 - Antonio Montes				No	0	0	0	--	--	--	
2003 - Joseph Montes				No	0	0	0	--	--	---	
2004 - Miguel Rivera				No	0	0	0	--	--	---	
2005 - Oscar Romero				No	0	0	0	--	--	---	
2006 - Maynor Peralta				No	0	0	0	--	--	--	
2007 - Roger Gaitan				No	0	0	0	--	--	--	
2009 - Sergio Martinez				No	0	0	0	--	--	--	

Below we will describe each of the columns displayed in the Widget.

Extensions, Extension Number and Name.

Reg Status, Green à Registered, Red à Not registered.

DND, Green à The extension is not available. Gray à The extension is available.

Q. Status, Green à The extension is registered in the queue. Red à the extension is not registered in any queue, Red Man with line à The extension does not belong to any queue.

In Call, Yes à the extension is on a call. No à the extension is free.

N. Out Calls, Number of outgoing calls on the current day.

N. In Calls, Number of incoming calls on the current day.

Total Calls, Number of outgoing and incoming calls on the current day.

Last Out Call, Shows how long ago the last outgoing call of the current day was from the extension.

Last In Call, Shows how long ago the last incoming call was for the current day from the extension.

Pause Reason, Shows the reason for the extension’s last pause.

Details, Shows details of incoming, outgoing and internal calls of the extension.

Administration

Users

You can create multiple users associated with your own extension. Each user can have certain privileges regulated by the Roles to which he is associated. To configure users, go to Administration/Users and create Users.

GENERAL

Full Name *

Username *

E-mail

Password *

Repeat Password *

Role

Administrator

Tenant ⓘ

Attimo

Layout

Default

Assigned Extension ⓘ

None

Supervisor's Extension ⓘ

None

Timezone

(GMT +0:00) UTC

Language

English

Queues Sound Notification ⓘ

☐

Active Account ⓘ

☒

Fill the following information:

Full name, full name of the person.

Username, username to login to the application.

E-mail, Email associated with the user.

Password, password to login to the application.

Repeat Password, confirm password to login to the application.

Role, select the role (created previously).

Tenant, allows you to define to which tenant belongs to this user. When this user logged into to the platform, it will be able to only see info for the tenant assigned.

Layout, select Layout to show it in startup.

Assigned Extension, the user can use this extension to login to the system or execute any allowed actions.

Supervisor's Extension, the users with a supervisor assigned will be able to request help to its supervisor from the.

Time Zone, time zone of the user.

Language, language to show the interface.

Queue Sound Notification, if is checked, a sound notification is played when a new caller joins to any queue visible in your switchboard.

Activate Account, if you un-check this option the user won't be able to login to the GUI.

Roles

Roles are the rights that each user has. To configure it you must go to Roles and select the option that you want the Role to have access to.

Fill the following information:

Update Profile, If checked the users with this role assigned will be capable to configures its own user settings.

Share Layouts, Defines if the users with this role will be able to share its layouts with other users.

Layouts Creation, If checked, the users with this role assigned will be able to create its own layouts.

Default layout, If checked, the users with this role assigned will be able to access to the default layout.

Agent Mode, If checked the users with this role won't be able to execute any actions on the widgets except on their own extension.

Menu Access:

Switchboard, Gives access to the Switchboard, seeing PBX activity in real time.

Users, Gives permissions to create users.

Roles, Gives permissions to create Roles for users.

Settings, Gives permissions to create Pause Causes and modify the menu context trigger.

Licensing, Gives permissions to activate/revoke or view license status.

Then in the Privileges, we have two options. One, the items included in the Role, and the second include Actions.

Next, we are going to select the different privileges that the user will have at the level of Extensions, Queues parking lots, Conference and Trunks.

Extensions, Allow you to define the allowed Extensions by this role.

Queues, Allow you to define the allowed Queues> by this role.

Parking Lots, Allow you to define the allowed Parking Lots by this role.

Conferences, Allow you to define the allowed Conferences by this role.

Trunks, Allow you to define the allowed Trunks by this role.

In Extensions Actions, we have:

Extensions Actions, Allow you to define the allowed Extensions actions by this role.

- Originate Call
- Spy
- Spy with Whisper
- Transfer
- Supervise Transfer

In Queue Actions, Allow you to define the allowed Queues actions by this role:

- Add Member
- Remove Member
- Pause/Unpause Agent

In Parking Action, Allow you to define the allowed Parking Lots actions by this role:

- Un-park

In Conferences Actions, Allow you to define the allowed Conferences actions by this role:

- Kick
- Mute/Un-Mute

In Queue Calls Action, Allow you to define the allowed Queued Calls actions by this role:

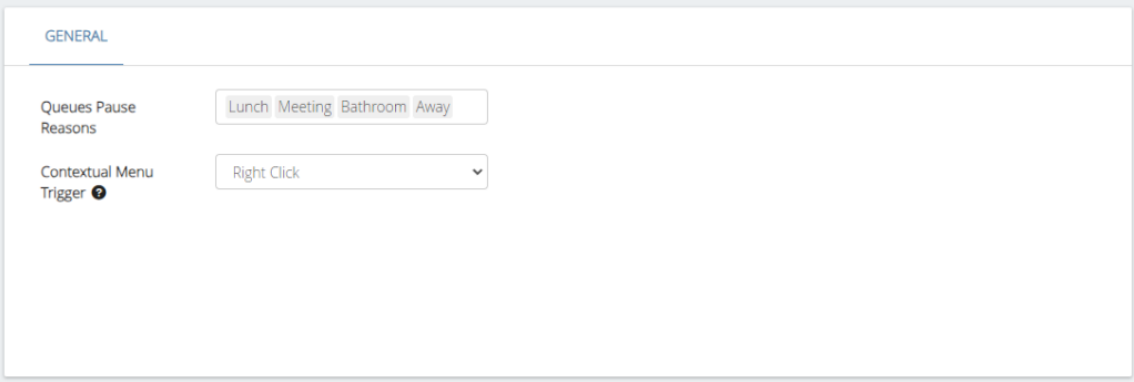
- Take Call
- Transfer Calls

In Trunk Actions, Allow you to define the allowed Trunk actions by this role.

- Hangup

Settings

In Settings the reasons of Pause are configured, very useful when creating reports by agents. To configure this option, go to Switchboard/Administration Settings.



The screenshot displays the 'GENERAL' settings section. Under 'Queues Pause Reasons', there is a multi-select box with the following options: Lunch, Meeting, Bathroom, and Away. Below this, the 'Contextual Menu Trigger' is set to 'Right Click' via a dropdown menu. A small help icon is visible next to the trigger label.

Fill the following information:

Queues pause Reasons, add the different pause reasons.

Contextual menu Trigger, specifies what event on widgets elements triggers the context menu.

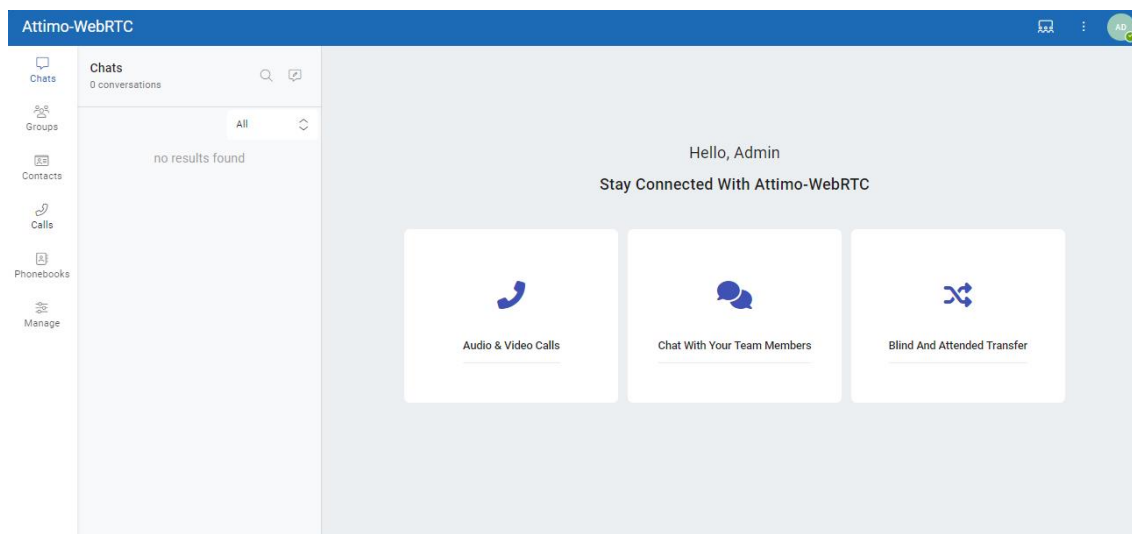


WebRTC

INDEX

Menu.....	3
Groups	5
Contacts	6
Calls	7
Menage	10
Settings.....	15

Menu



On the menu we got the following options:

Chats, list of people with which the user has had some type of chat.

Groups, List of available groups with their respective members.

Contacts, the contact list added by the user manually. To enable/disable BLF for a number, you should remove the number and then add it again.

Calls, The user's Dialpad and call history.

Phonebooks, phone directories that are extracted from the Attimo CloudPBX Phonebook.

Manage, main menu to configure the system.

Users, Create and edit VitXi Users.

Role Manager, Create and edit privilege profiles for users.

Groups, Create and edit groups to which users belong.

Agent Profile, Create and edit profiles for call center agents.

BLF Settings, Manage the number of BLF contacts allowed per user to prevent server overload and optimize performance. Set a value between 1 and 100.

Presence Status, Create and edit the possible states of user presences.

Social Networks, Configure the URL of each Tenant's social networks.

Mail Settings, Configure the email account to send emails.

ICE Server, Configure the ICE Server. We recommend leaving the default ones.

PBX Settings, Configure Attimo CloudPBX location. It is only necessary to configure when you have a connection problem or want to use a Attimo CloudPBX that is on another server.

-  Start an audio conference.

Audio Conference

Users

Contacts

Participants (0) * Up To 10 Participants.

No users available

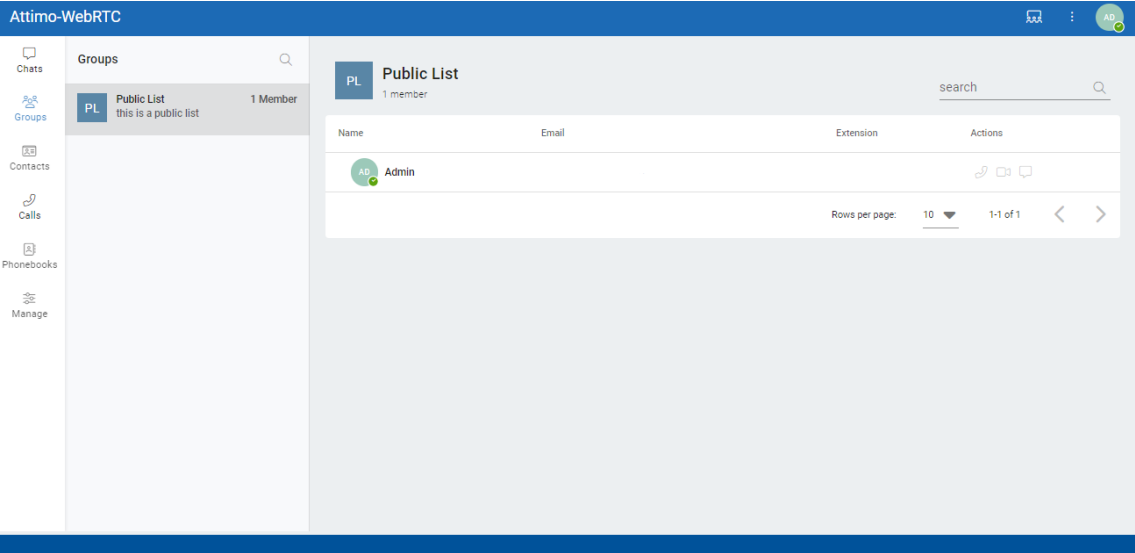
 START CONFERENCE

CANCEL

Groups

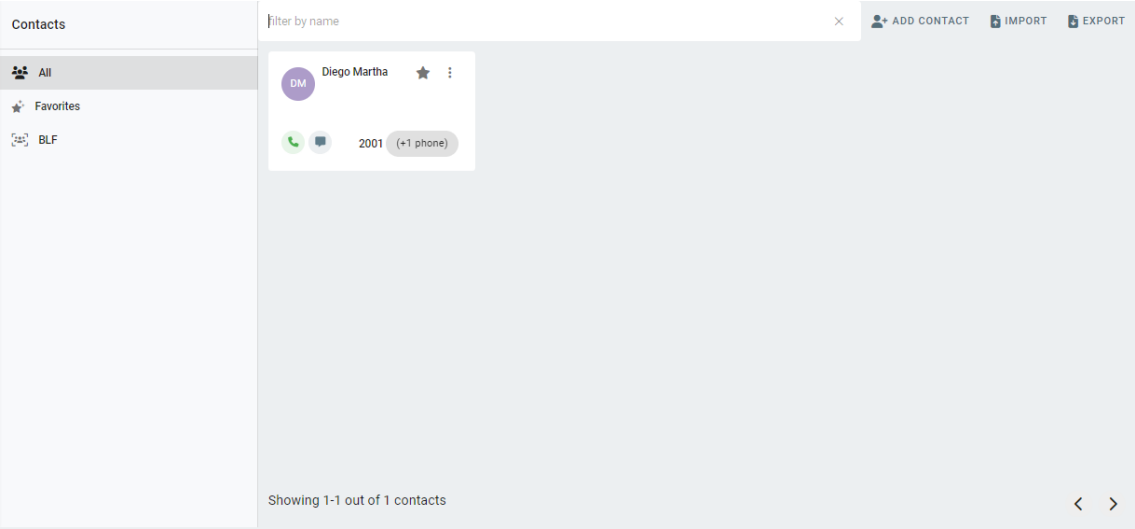
It is possible for group users so that they can be accessed only by the members of their group, thus the identity of the different sectors of a Company can be preserved.

With this option it is possible to see the members of the groups to which the user has access.



Contacts

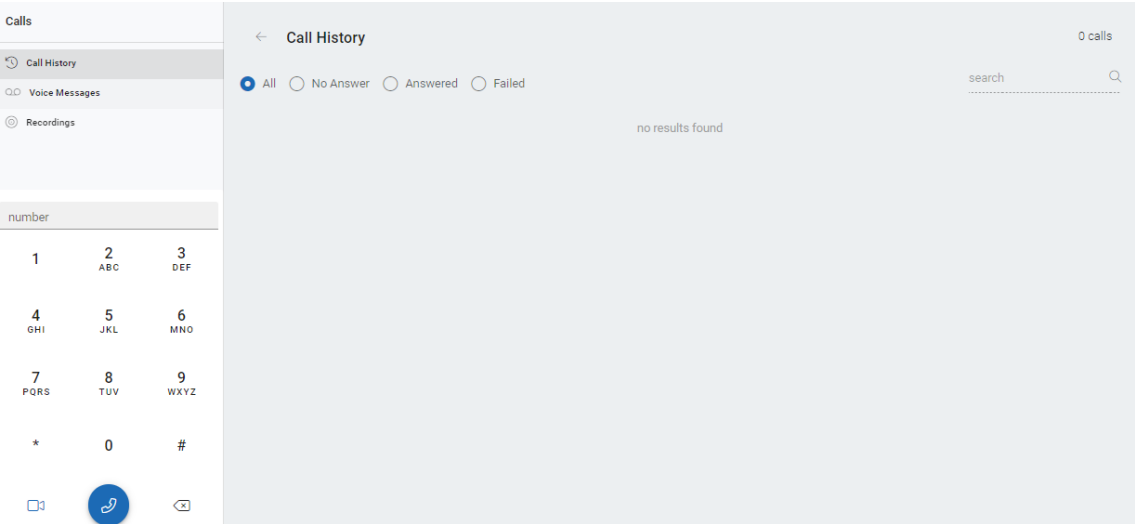
It is possible to save contacts through this tab; it is also possible to add, import, and export them. It is also possible to filter by the name that was saved.



Calls

Call History

To see the call history, we must go to the Calls icon and press Call History. There we can see the last calls, for performance this history is limited by default to 250, however you can modify it.

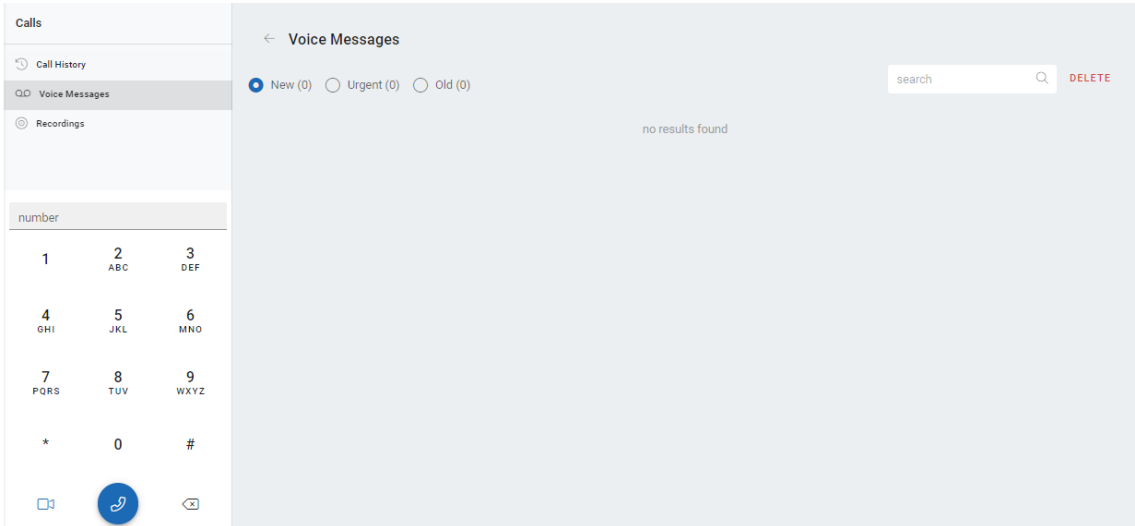


Voice Messages

It is possible to view all your Voicemail messages from VitXi WebRTC, which allows us to perform the following actions:

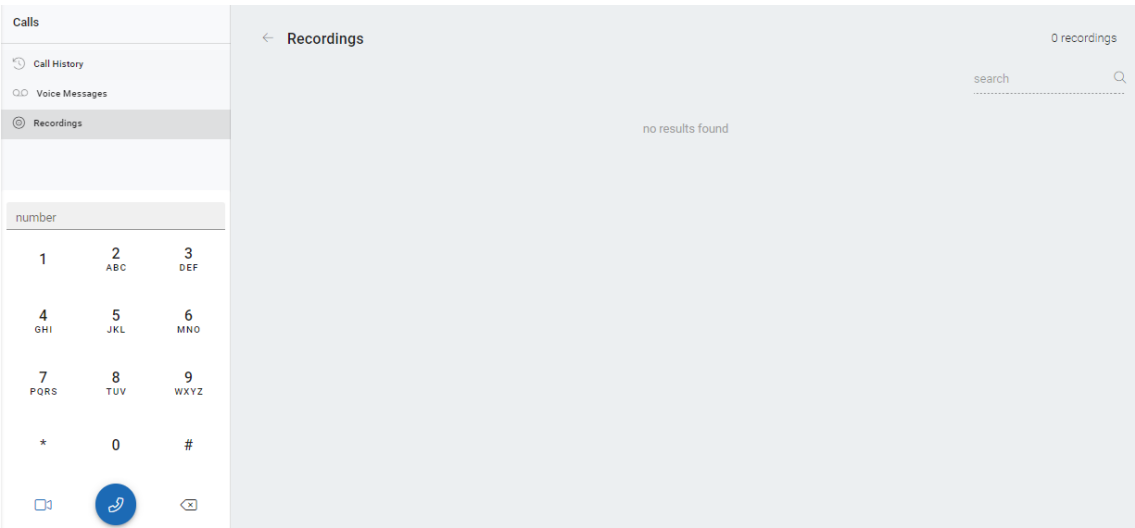
Listen.

Delete.



Recordings

It is possible to listen to the user’s call recordings from the extension they have associated to them.



Placing Calls

There are many ways to perform calls through Attimo WebRTC. The easiest way is by selecting the user and then pressing one of the three options available:

-  , Audio/Video Call.
-  , Chat Message.

We can also use the Number Pad by pressing the “Calls” button.



When a call is established, the following screen is displayed.

When there is an established call, the following options are displayed, which we will describe below.



Mute the microphone by selecting a new one from a list.



Enable/disable video casting or select a video source from a list.



Share screen.



Keyboard for sending DTMF tones.



Put the call on hold.



Call transfer.



Add a new call to the call to put it in conference.



Hang up on the current call.

Menage

Users

Now, we will create the users and assign them to the groups we want to. A user can belong to one or more groups.

The screenshot shows the 'User manager' interface in Attimo-WebRTC. On the left is a sidebar with navigation options: Chats, Groups, Contacts, Calls, Phonebooks, and Manage (highlighted). Under 'Manage', there are sub-options: Users, Role Manager, Groups, Agent Profiles, BLF Settings, Presence Status, Social Networks, Mail Settings, ICE Servers, and PBX Settings. The main area is titled 'User manager' and contains a 'Create' form. The form fields are: name, username, email, password (with a strength indicator), role (set to 'User'), tenant (set to 'Attimo (main tenant)'), groups (set to 'public list'), extension, agent profile (set to 'Default Profile'), and phonebooks. A 'Save' button is at the bottom right. A note at the bottom right states: '* assign an agent profile to be used in case the assigned role has call center permissions.'

Name, user's full name.

Username, username with which the user will log into Attimo WebRTC.

Email, the email address associated with the user, is very useful to recover the password.

Password, password with which the user will log into Attimo WebRTC.

Role, the type of user. There are three types of users.

- **Super Administrator**: capable of managing all the users and tenants.
- **Administrator**: capable of managing users within its tenant.
- **User**: Attimo WebRTC end-users.
- **Agent**: When it is a Call Center Agent, we recommend this profile already activates an icon in the interface to facilitate the Login/Logout, Pause/Unpause of the Agent in queues.

Tenant, tenant to which the user belongs to.

Groups, groups to which the user belongs to.

Extension, extension associated to the user. For this, the user needs to have a Attimo Device created on Attimo CloudPBX.

Agent Profile, Assigning an agent profile to be used in case the assigned role has call center permission.

Phonebooks, phonebooks to which the user will have access to.

Role Manager

To facilitate the creation of users with their respective permissions, roles are created that are assigned to different types of users.

←

Role manager

i

Create

≡

Name

Permissions

☐ Miscellaneous

☐ View Recordings i

☐ Update Social Networks i

☐ Management

☐ Users i

☐ Agent Profiles i

☐ Call Center

☐ Call Center Features i

☐ Queue Selection i

SAVE

Groups

Groups are created here.

←

Groups

i

Create

≡

name

description

tenant

Attimo - (main tenant)

▼

Save

Name, names of the groups.

Description, description given to the groups.

Tenant, tenant to which the group belongs to.

Agent Profiles

When we are using Call Center Agents in Vitxi and we want each agent group to have a different profile in terms of types of breaks or integration with CRM, we must create profiles with this information.

Below we show the interface where these profiles are created.

The screenshot shows the 'Agent profiles' creation interface. At the top, there is a header bar with a back arrow, the title 'Agent profiles', and an information icon. Below the header, there is a 'Create' button and a hamburger menu icon. The main form area contains several fields and options:

- Name:** A text input field for the profile name.
- Is Global:** A checkbox with the label 'Is Global'. Below it, a note states: '*If Checked, The Profile Can Be Occupied By Any Of The Tenants.'
- Tenant:** A dropdown menu currently showing 'Attimo'.
- CRM URL (Optional):** A checkbox with the label 'CRM URL (Optional)'. Below it, a note states: 'Configure An Endpoint Of Your CRM To Save Data From A Call.' There is a 'base URL' field with a placeholder 'https://your-crm.com'.
- Custom Website Tab:** A checkbox with the label 'Custom Website Tab'. Below it, a note states: 'Add A Custom Tab To The App With An Embedded Website'. There is a 'tab name' field.
- Dynamic Variables:** A section with a 'web url' field and a note: '* enter a secure web site (https://) that you can trust. this is important to keep your users safe.' Above this field, there are labels for 'Parameter Name' and 'Value', and a checkbox for 'Use Dynamic Variable'.
- Reasons For Pauses:** A section with the title 'Reasons For Pauses' and a subtitle 'Pause Reasons Used When An Agent Pauses A Session In Queues'. There is an 'Add' button.
- Pause:** A text input field for defining pause reasons.

Name, Name of the profile to create.

Is Global, if it is going to be shared to be used by all Tenants.

CRM URL, you can set the URL/endpoint of your CRM. You can configure dynamic or static variables. The CRM URL will open when the agents click the CRM button in the call layout.

Custom Website Tab, configure a URL/website to be inserted as a “custom tab.” This way, agents will be able to visit the site without leaving Vitxi.

Reason for Pauses, we define all possible reasons for pauses.

BFL Settings

Manage the number of BLF contacts allowed per user to prevent server overload and optimize performance. Set a value between 1 and 100.

The screenshot shows the 'BLF Contact Control' settings interface. It has a title 'BLF Contact Control' with a person icon. Below the title, there is a description: 'Manage the number of BLF contacts allowed per user to prevent server overload and optimize performance. Set a value between 1 and 100.' There is a 'Limit' field with a value of '50'. At the bottom right, there is a green 'Save' button.

Social Networks

Here the URLs of the social networks are configured. This can be configured by Tenants. When they are configured, some icons will appear at the bottom right.

←

Social networks

Tenant

Attimo

https://facebook.com/ attimo

Set the username of your facebook page

https://twitter.com/ attimo

Set the username for your twitter account

https://youtube.com/ attimo

Set the username of your youtube channel

https://instagram.com/ attimo

Set the username for your Instagram account

Update

Mail Settings

Configuration of the email account to be able to send email in case some users need to reset their password.

Mail settings

smtp

host

smtp.office365.com

port

587

username

required

password

tls

from address

required

from name

Attimo Web RTC

Send email test

Save

ICE Servers

←

Ice servers

i

Servers

Server - 1

STUN or TURN URI (1)

stun:stun1.l.google.com:19302

STUN or TURN URI (2)

stun:stun2.l.google.com:19305

+

−

☐

Authentication

+ ADD SECONDARY SERVER

Save

PBX Settings

Even though this information is asked at the moment of installation, it is also possible to perform some modifications afterwards.

←

Pbx settings

i

Settings

Connected

host

localhost

https port

443

VitalPBX HTTPS port (useful for obtaining assets from the branding module).

PJSIP Port

50702

Configure the port to be used for extension registration

WSS port

8089

If you have a firewall, don't forget to allow TCP port through so your client can connect.

API port

3501

Indicate the port used by the API.

☐

Edit application key

Save

Settings

The first thing that we are going to configure is the System Settings, for which we will go to the main menu and select settings. This will be displayed in the following screen.

Profile

Shows the name, Username and extension number of the user.

Settings

PROFILE

ACCOUNT

GENERAL

AUDIO

HEADSET
INTEGRATION

VIDEO

CALLS

CALL CENTER

Profile

 Full Name
Admin

 Username
Admin

 Extension

Account

Change the user’s password.

Settings

PROFILE

ACCOUNT

GENERAL

AUDIO

HEADSET
INTEGRATION

VIDEO

CALLS

CALL CENTER

Account

 Change Password

General

Timezone, the time zone for the user. This way, the user will see their own time zone, regardless of their location.

Desktop Notifications, the user can receive a notification for certain events, for example, receiving a call, a message, and more.

Dark Mode, here you can select the dark mode view of the interface.

Settings

PROFILE

General

Timezone

America/Sao_Paulo

ACCOUNT

GENERAL

☒

Desktop Notifications

Receive Desktop Notifications For Certain Events

AUDIO

HEADSET INTEGRATION

☐

Dark Mode

Change The Appearance Of Attimo-WebRTC To Dark Tones

VIDEO

CALLS

CALL CENTER

Save

Audio

Microphone, Select the microphone to use.

Audio, Select the audio output to listen to.

Ringling Device, Select the audio output to hear the ring when a call comes in.

Echo Cancellation, this is used to prevent the echo effect during a call.

Noise Suppression, automatically filters and eliminates background noise.

Media Volume, Adjust the volume to hear the ringtone when a call or notification comes in.

Settings

PROFILE

ACCOUNT

GENERAL

AUDIO

HEADSET
INTEGRATION

VIDEO

CALLS

CALL CENTER

Audio

Microphone

Padrão - Microfone (Linkvil H301-U) (3391:3009)

Audio

Padrão - Fones De Ouvido (Linkvil H301-U) (3391:3009)

Ringling Device

Padrão - Fones De Ouvido (Linkvil H301-U) (3391:3009)

☒

Echo Cancellation

Feature Which Attempts To Prevent Echo Effects On A Two-Way Audio Connection

☐

Noise Suppression

Automatically Filters The Audio To Remove Background Noise

Media Volume

This Configuration Will Be Applied To The Ringtones Of Calls And Other Sounds Of The Application.

Save

Headset Integration

HID enablement module for streamlined call actions using headsets. Enhance the calling experience with one-touch controls, including call answering and volume adjustments.

Settings

PROFILE

ACCOUNT

GENERAL

AUDIO

HEADSET INTEGRATION

VIDEO

CALLS

CALL CENTER

External Headset/Headset Integration

☐ HID Integration

Save

Video

Camera, Select the camera to use.

Resize Mode, Specify the resolution mode of the video track.

Settings

PROFILE

ACCOUNT

GENERAL

AUDIO

HEADSET INTEGRATION

VIDEO

CALLS

CALL CENTER

Video

Required Track Is Mis

Camera

Resize Mode
Specify The Resolution Mode Of The Video Track.

None

Save

Calls

Ringtone, here you can select the ring tone to listen to when the user is receiving a call.

Auto Answer, this will answer the call automatically.

Call history limit, with this you can establish the call limit to show on the Calls option .

Settings 

PROFILE

Calls

ACCOUNT

Ringtone

Ringtone to be played on incoming calls



Telephone Ring Variant 1



GENERAL

☐

Auto Answer

AUDIO

HEADSET INTEGRATION

Call history limit

Set the limit of records to show in call history

250



VIDEO

Save

CALLS

CALL CENTER

Call Center

Automate Agent Login and Logout when starting and closing VitXi. CRM integration options are also configured to display pop-up windows.

Settings 

PROFILE

Queues

ACCOUNT

☐

Log Into Queues On Startup

Automatically log in to assigned queues when starting the application

GENERAL

☒

Log Out From Queues When Closing The Application

AUDIO

HEADSET INTEGRATION

CRM URL

Open The CRM Url In:

In case your agent profile has a crm url configured, choose how the url should be opened in your browser.

☒ New Tab

☐ New Window

(This option is subject to browser and operating system permissions.)

VIDEO

CALLS

CALL CENTER

☐

Open CRM Url Automatically

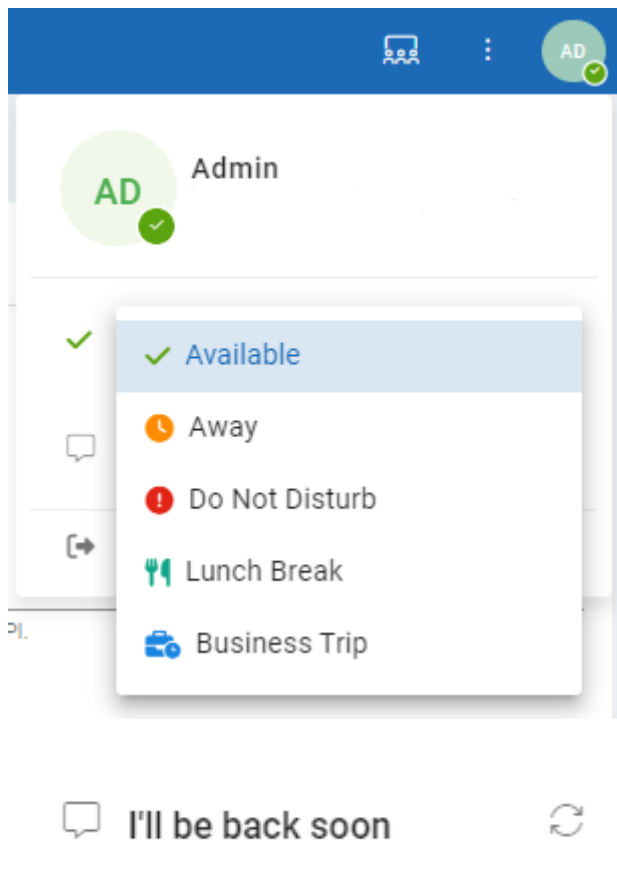
The crm url will open automatically when the call is established.

Save

Presence

Presence Status, select the presence status with which we desire to have when login in.

Additional Information, if the user wants to show some additional message that other users that are connected will see.



Here, the system administrator can create the different presence status that users can use. By default, there are five (5) statuses which are, Available, Do Not Disturb, Lunch Break, and Business Trip, however, you can add more.

