1. Introduction

The VoIP (Voice over Internet Protocol) telephone system is a system that allows making telephone calls over the internet. Instead of using traditional telephone lines, voice communication is transmitted through data packets over the internet.

A VoIP telephone system can be implemented both in business and residential settings. It consists of an IP telephony server that manages the calls and related resources. The phones connected to the VoIP system can be specific VoIP hardware devices or softphones, which are phone applications installed on computers, smartphones, or tablets.

The advantage of using a VoIP telephone system is that it offers advanced communication features and can be more cost-effective than traditional telephone lines. Some common features include voicemail, call forwarding, call waiting, conference calling, and integration with other communication applications such as email and instant messaging.

Furthermore, the VoIP telephone system allows the flexibility to make and receive calls from anywhere with an internet connection, which is particularly useful for businesses with remote employees or multiple offices in different geographic locations.

However, it is important to emphasize that a stable and quality internet connection is crucial to ensure the quality of VoIP phone calls.

2. Technology

The telephone system is built using a screen structure with the software called PHPRunner, and the code and project are available on GitHub along with the software's ISO.

The application layer is handled by PHP along with Asterisk as the call handling middleware. Asterisk is an open-source software application that acts as middleware for call control and VoIP. It provides a complete platform for building IP-based communication systems, including VoIP telephone systems, conference servers, IVR (Interactive Voice Response), and more.

One of the key advantages of Asterisk is the flexibility and customization it offers. Being open-source software, developers can modify and adapt Asterisk to meet the specific needs of an organization. Additionally, there are many modules and additional features available in the Asterisk community, allowing further expansion of its functionalities.

In summary, Asterisk is a middleware application that plays a crucial role in call control and the implementation of IP telephony and VoIP solutions.

The telephone system is prepared for integration with several technologies, such as FOP2 (Flash Operator Panel 2 - https://www.fop2.com/), which can be acquired with an additional license to have more call controls. To activate its functionality, you just need to acquire and add the software license.

The G729 codec can also be acquired directly from the Asterisk developers (https://www.asterisk.org/products/add-ons/g729-codec/). This codec aims to compress packets and use less network traffic.

Another component that can be added to the telephone system is the Digium fax, which can be acquired and added to the server for compatibility with the T.38 standard. For use in alaw networks or with Khomp equipment (physical devices), the fax component that comes with the package supports fax reception and creation of PDF files.

3. Requirements

In terms of hardware requirements, a minimum of a 3 GHz x86 processor, 1 GB of RAM, and 30 GB of disk space are required. However, in production environments, it is recommended to consider factors such as recording retention time, CPU and memory resource consumption, as well as SLA calculations, the number of VoIP clients, and/or the number of open screens querying the telephone system. These factors may require an increase in resources based on the number of users involved.

It's important to assess the specific needs and demands of your environment to determine the appropriate hardware specifications for optimal performance and scalability.

4. Installation

To install the telephone system, follow these steps:

- Download the VoIP system ISO and create a DVD or start a virtual machine with the ISO.
- During the machine's boot, press the <Enter> key to be guided through the installation process and receive information about the progress.
- After the installation is complete and the computer restarts, you will be presented with a command prompt. Access it using the "root" username and the password "_cell".
- Once you access the command prompt, type "setup" and follow the default configuration procedures for CentOS 5.5.
- After completing the network configuration, access the telephony website using the server's IP address (http://ipoftheserver). The default username is "admin", and the password is "_cell".
- It is recommended to start the configuration on the parameter screen, where you will enter the starting and ending extensions to set up the necessary extension range.
- Proceed to configure the outbound and inbound call interfaces according to your needs.
- Continue configuring the auto-attendant menus, call queues, agents, and other options detailed in the following chapters, according to your company's specific requirements.

Make sure to carefully follow the instructions and configure the options according to your needs to ensure the proper functioning of the telephone system.

Note: If the server is accessible over the internet, it is highly recommended to set up a firewall and use the security script provided with the IPBX installation ISO. These measures help ensure the system's protection against potential attacks and secure data access to the company's dialing system.

Information described below are descriptions of what you can do screen by screen:

5. IPBX description

5.1. Painel de operações (Operation Panel)

Panel where all configured extensions in the system are listed. To access the "Painel de operações", it is necessary to acquire a license for FOP2 (https://www.fop2.com/). The IPBX system is fully compatible with FOP2 version (2-2.26), which is already installed with the telephony system ISO, only requiring the license to be obtained.

With the operations panel, you will have access to buttons that can:

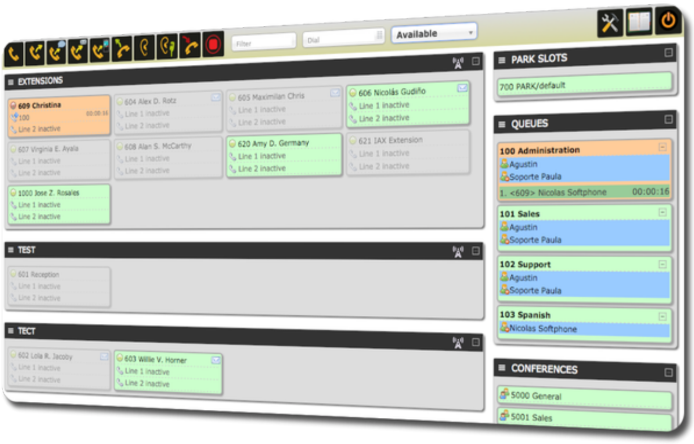
- Answer/End Call Button: Allows the operator to answer an incoming call or end an active call.
- Transfer Button: Used to transfer a call to another extension or phone number.
- Speed Dial Button: Enables quick dialing of a pre-configured number to make a call.
- Mute Button: Pressing this button allows the operator to mute the microphone audio during a call.
- Hold Button: Used to put a call on hold. The button can display the status of the call on hold, allowing the operator to resume the call later.
- Call Pickup Button: Allows the operator to pick up a ringing call from another extension.
- Monitoring Button: Enables the operator to monitor a specific extension to view real-time status of active or waiting calls. It is also possible to monitor a call in whisper mode, where you can speak to the operator without the other party hearing.
- Call Parking Button: Used to park a call in a specific position for later retrieval from another extension.
- Call Recording Button: When activated, starts recording the ongoing call.
- Volume Control Button: Allows adjusting the speaker or headset volume during a call.

There are also panels that provide an overview of your system:

- Extension Status Panels: These panels show the status of the extensions (lines) in the telephony system. This includes information about extensions in use, available, busy, or on hold.
- Active Calls Panels: These panels display ongoing phone calls, showing the source and destination numbers, call duration, and other relevant information. This allows the operator to quickly visualize active calls.
- Waiting Calls Panels: If there are calls waiting in the system, FOP2 panels display information about these calls, such as the waiting call number and the extension it is directed to.
- Caller ID Display: When available, FOP2 can show caller identification information, such as the contact's name or caller's phone number.
- Recording Status: If call recordings are in progress, FOP2 panels can show the status of the recordings, indicating which calls are being recorded.

- Wait Time Information: In systems that support call queues, FOP2 can display information about the average wait time for queued calls and the number of calls awaiting assistance.

Here's an example of the screen:



5.2. Configurações Pessoais (personal configuration)

In the configuration of your user account, which you used to authenticate yourself in the system, you can access a screen where you can change the password for your extension/user and the personal name associated with your account. Additionally, there are other options available, such as configuring call forwarding for calls originated externally to the company (DID) or forwarding all calls, including to a mobile phone, another extension, or an external phone.

You can also perform transfers with predefined schedules set by the administrator, configured as call forwarding. You have the option to add a lock to your extension (lock feature). Furthermore, you can configure it so that before a call is actually made, it is necessary to enter the cost center that will be accounted for in the call (through an enable/disable option). Another available configuration is call overflow to another extension or phone system.

Here's an example of the screen:

Ramal	7700
Senha	13219444 *
Nome	OPERATOR
E-mail	
Entrada no DDR desvia para:	Favor Selecionar
Qualquer ligação desvia para:	
Em horários e dias específicos desvia para:	Favor Selecionar
Trava de ramal (cadeado)	☑
Informar centro de custo	☐
Transbordo	Toca 30 segundos, desvia para:

5.3. Central Telefônica

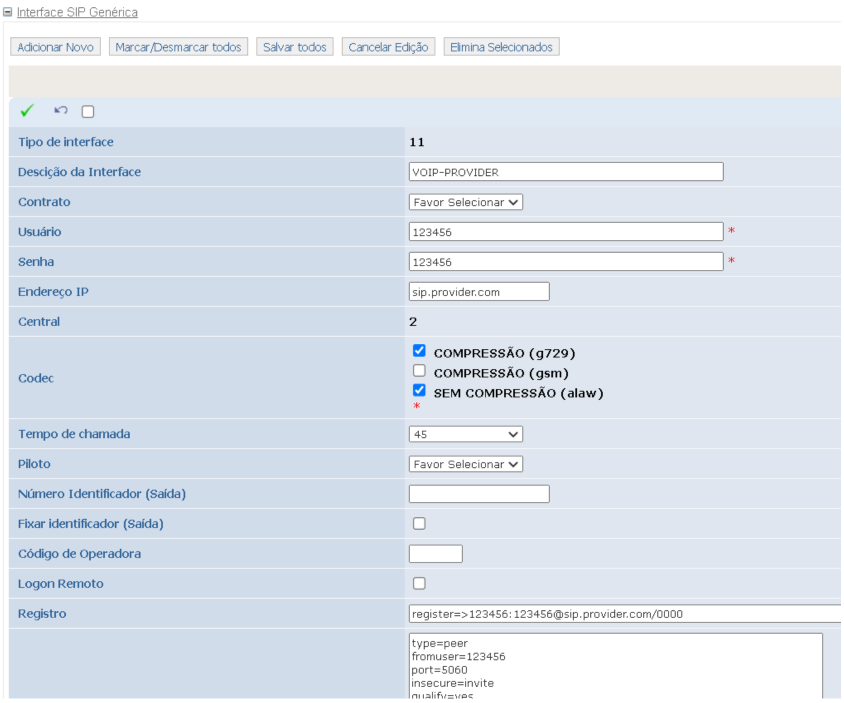
5.3.1. Interfaces

The “interfaces” screen is where the configuration for integration with call inputs and outputs is located, through SIP accounts, MS-LYNC, CISCO CALLMANAGER, or Khomp devices, which include (E1, FXO, GSM, FXS).

In this screen, you can configure the type of compression to be used, the call timeout duration, and fix the source phone for the entire number range.

Additionally, you can configure the contract that will be used to calculate the cost of calls, define the carrier code, and determine how incoming and outgoing calls will be routed through the interface.

Here's an example of the screen:



5.3.2. Plano de Discagem (Dial Plan)

The “plano de discagem” / dial plan is used to determine how a dialed phone number will be interpreted and routed by the system. It is designed to provide instructions to the system on how to handle different patterns of phone numbers and route them to the correct destination.

Some of the functionalities and configurations that can be set in a dial plan include:

Number Format: Defines how phone numbers should be dialed, according to the call output configuration set in the interface screen.

Call Routing: Specifies how calls are directed based on the dialed numbers. This can include defining specific routes for different types of numbers, such as local, national, or international calls.

Numbering Rules: Sets rules for checking and validating the dialed phone numbers, ensuring they are correct and following the expected pattern.

The dial plan is an essential part of configuring a telephone system, especially in IP-based systems like Asterisk. It helps ensure the proper rotation and routing of calls according to the needs and policies of an organization.

Here's an example of the screen:

Após realizar as alterações, salve e aplique as configurações atraves do botão abaixo.

Aplicar Configurações

Adicionar Novo | Salvar todos | Cancelar Edição | Elimina Selecionados

Padrão	Contexto	Rota	Interface Saída	Adiciona na Rota	Interface Contingência	Adiciona na Rota
[1-6]XXXXXXX *	FIXO-LOCAL *		VOIP-PROVIDER		Favor Selecionar	
[7-9]XXXXXXX *	CELULAR-LOCAL *		VOIP-PROVIDER		Favor Selecionar	
XX[7-9]XXXXXXX *	CELULAR-DDD *		VOIP-PROVIDER		Favor Selecionar	
XX9XXXXXXXX *	CELULAR-DDD *		VOIP-PROVIDER		Favor Selecionar	
XX[1-6]XXXXXXX *	FIXO-DDD *	0	VOIP-PROVIDER		Favor Selecionar	
XXXXXXXXX. *	DDI *	00	VOIP-PROVIDER		Favor Selecionar	
0800. *	PBX-RAMAL *		VOIP-PROVIDER		Favor Selecionar	
XXX *	PBX-RAMAL *		VOIP-PROVIDER		Favor Selecionar	
0300. *	PBX-RAMAL *		VOIP-PROVIDER		Favor Selecionar	
XX[1-6]XXXXXXX *	FIXO-DDD *		VOIP-PROVIDER		Favor Selecionar	
XX9XXXXXXXX *	CELULAR-DDD *	0	VOIP-PROVIDER		Favor Selecionar	
XX[7-9]XXXXXXX *	CELULAR-DDD *	0	VOIP-PROVIDER		Favor Selecionar	

It is important to note that if you have different outbound options, you can configure, for example, calls to mobile phones using a GSM interface or even another contracted VoIP plan. Additionally, in case there is any issue with the call output, you can use a second channel through the contingency interface, which can be of any type.

For this reason, we recommend not adding the carrier code in this field. It should be entered in the corresponding field for the interface. This way, if there is any problem with the default interface, the telephone system will make calls using the contingency interface with a different carrier code if necessary.

Other route controls and formatting options can be viewed in the example shown in the figure.

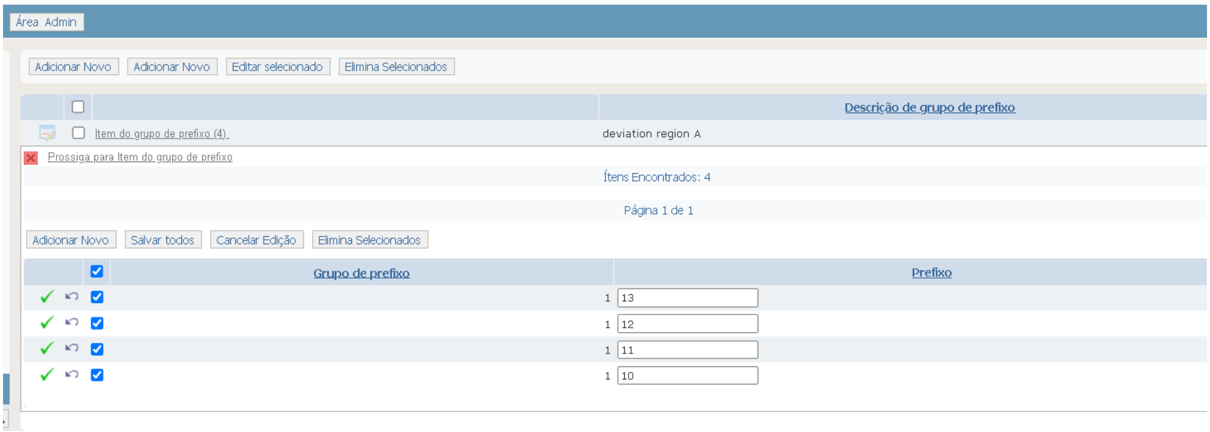
5.3.3. Menu de Atendimento » Desvio por prefixo (Deviation by prefix)

When a call is received on the interface (as mentioned earlier), it is possible to configure a pilot extension, which can be a menu, for example.

In the following example, I am configuring a specific group called "A" that will match received phone numbers with the prefixes from 10 to 13. Let's assume you serve a particular region that has phone numbers with prefixes from 10 to 13. In this case, you may have a dedicated area within the company to handle calls from people in that region due to some affinity or efficiency in serving them.

This group of prefixes created will be used later in the call menu. It will serve as a criterion to route the call to the corresponding area of the company based on the dialed number's prefix.

Here's an example of the screen:

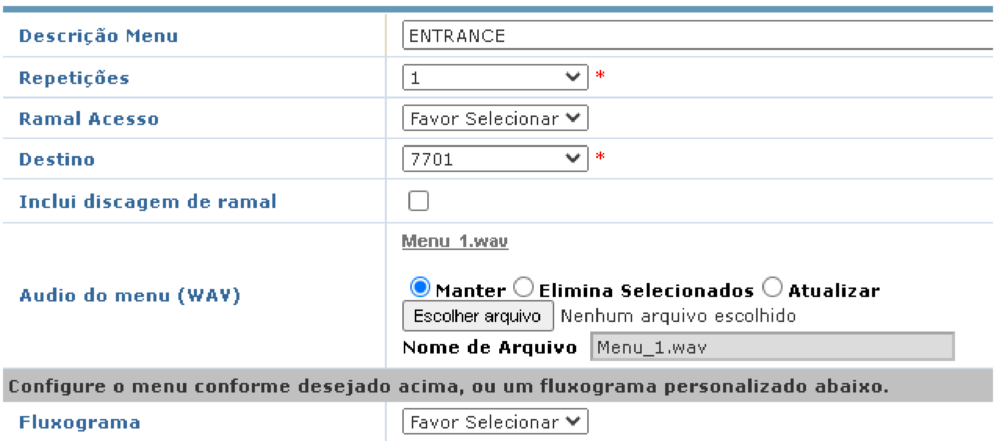


5.3.4. Menu de Atendimento (Menu attendance)

In the following example, we will present a configuration of a feature called IVR (Interactive Voice Response), which refers to the options you enter in the self-service menu. These options can be routed to a queue, an extension, or another call menu.

The configuration of the call menu consists of the following information: an audio prompt that will be played to the user, the number of times the audio will be repeated, the destination if no key is pressed, or if there are no matching rules for the user in the menu. A previously registered flowchart can also be used, which can be a SOAP query self-service system with a programmed call flow (this functionality will be detailed later).

This configuration allows users to interact with the self-service system through selected keypad options, directing them to different destinations based on their choices. It is a way to automate and route calls according to users' needs and preferences, providing more efficient and personalized service.



In the context related to the previous topic, in this section, the registration of prefix diversion is performed.

It is possible to configure an automatic diversion if the system identifies that the originating phone belongs to the prefix of the incoming call. In the following example, it checks if the phone has the registered prefix for region "A". If the call matches this prefix rule, it will be directed to the operator at extension 7700.

Furthermore, other diversion options can be observed, such as returning to the previous menu or sending the call to a service queue, among other available options on the screen. These diversions allow calls to be directed according to specific needs and rules defined for each prefix or situation, providing greater flexibility and customization in call routing within the telephone system.

Salvar todos | Cancelar Edição

			Prefixo	Digito	Ação do item de menu	Destino
✓	↶	☑	Favor Selecionar Adicionar Novo	0	Executa a acao *	Retorna Menu anterior *
✓	↶	☑	Favor Selecionar Adicionar Novo	1	Desvia para a fila *	SUPPORT *
✓	↶	☑	Favor Selecionar Adicionar Novo	2	Desvia para a ramal *	7700-OPERATOR *
✓	↶	☑	Favor Selecionar Adicionar Novo	3	Executa a acao *	Repete Menu *
✓	↶	☑	Favor Selecionar Adicionar Novo	4	Executa a acao *	Repete Menu *
✓	↶	☑	Favor Selecionar Adicionar Novo	5	Executa a acao *	Repete Menu *
✓	↶	☑	Favor Selecionar Adicionar Novo	6	Executa a acao *	Repete Menu *
✓	↶	☑	Favor Selecionar Adicionar Novo	7	Executa a acao *	Repete Menu *
✓	↶	☑	Favor Selecionar Adicionar Novo	8	Executa a acao *	Repete Menu *
✓	↶	☑	deviation region A Adicionar Novo	9	Desvia para a ramal *	7700-OPERATOR *

5.3.5. Ramais (Extension)

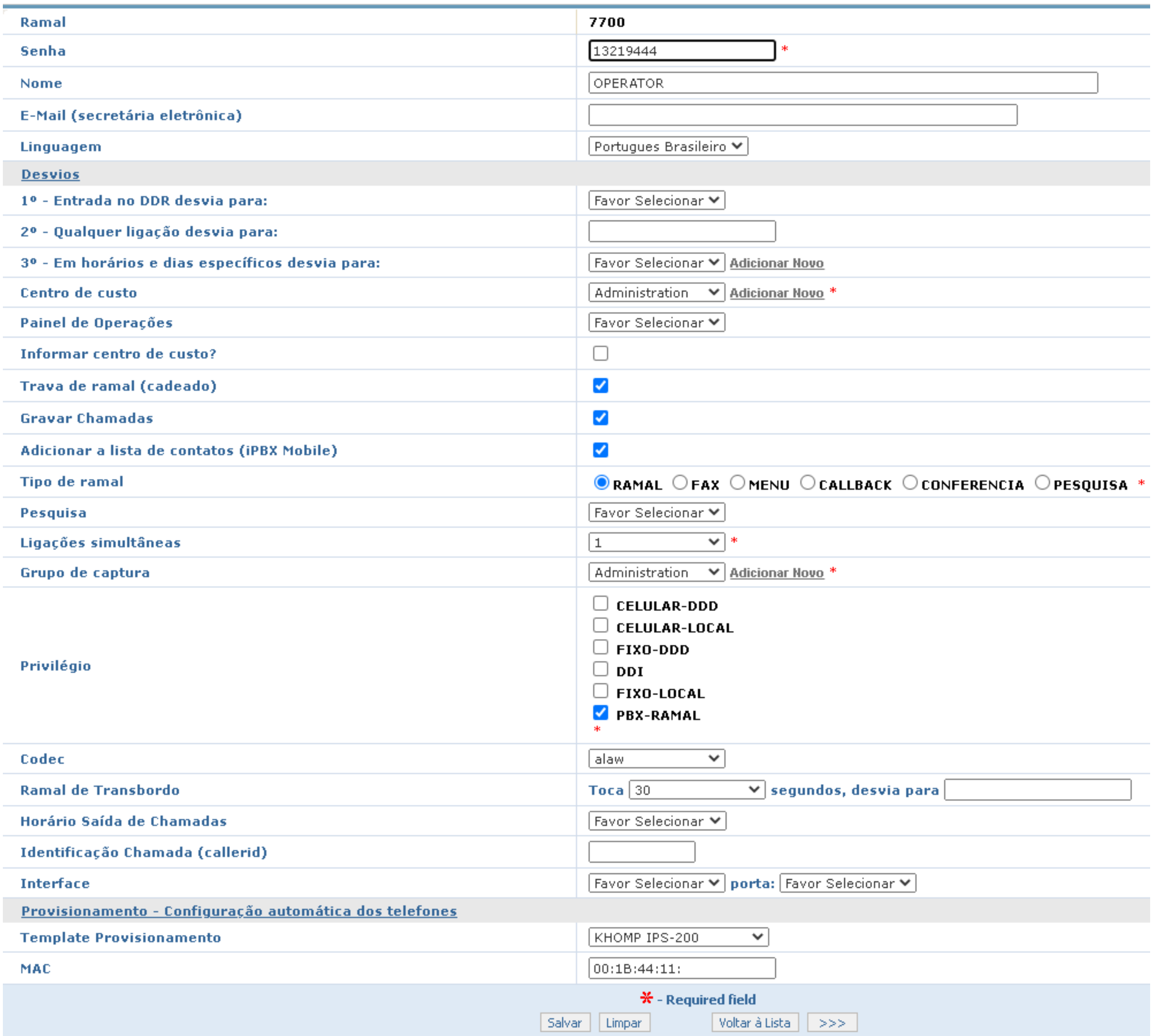
The configuration of extensions in the telephone system involves filling in the corresponding fields with the details of each extension.

These fields may include:

- Dialing privileges: Defines the dialing permissions for the extension, such as allowed numbers, specific restrictions, or blocks.
- Cost center: Allows assigning a cost center to the extension, which will be used to track the costs of calls made by the extension.
- Language: Sets the preferred language for messages and voice prompts related to the extension.
- Call forwarding: Allows configuring call forwarding for the extension, such as directing calls to another extension, external number, or voicemail if the extension is unavailable.
- Emails for voicemail messages: Specifies one or more email addresses where voicemail messages left on the extension will be forwarded.
- Maximum concurrent calls: Determines the maximum number of simultaneous calls the extension can handle.
- 100% VoIP phones for provisioning: Indicates whether the extension is provisioned for phones that support only VoIP calls (Voice over IP).
- Call capture group: Defines the group to which the extension belongs for call capturing purposes. This allows other extensions to capture calls ringing on this extension.

Additionally, it is possible to configure other specific information for the extension, such as whether the phone has been ported to another system and requires a diversion, the type of codec used for encoding and decoding call audio, among other additional options and settings that may vary depending on the telephone system used. These configurations allow customizing and adjusting the features and functionalities of each extension according to the organization's needs.

Example:



Ramal	7700
Senha	13219444 *
Nome	OPERATOR
E-Mail (secretária eletrônica)	
Linguagem	Portugues Brasileiro
Desvios	
1º - Entrada no DDR desvia para:	Favor Selecionar
2º - Qualquer ligação desvia para:	
3º - Em horários e dias específicos desvia para:	Favor Selecionar Adicionar Novo
Centro de custo	Administration Adicionar Novo *
Painel de Operações	Favor Selecionar
Informar centro de custo?	☐
Trava de ramal (cadeado)	☑
Gravar Chamadas	☑
Adicionar a lista de contatos (iPBX Mobile)	☑
Tipo de ramal	◉ RAMAL ○ FAX ○ MENU ○ CALLBACK ○ CONFERENCIA ○ PESQUISA *
Pesquisa	Favor Selecionar
Ligações simultâneas	1 *
Grupo de captura	Administration Adicionar Novo *
Privilégio	☐ CELULAR-DDD ☐ CELULAR-LOCAL ☐ FIXO-DDD ☐ DDI ☐ FIXO-LOCAL ☑ PBX-RAMAL *
Codec	alaw
Ramal de Transbordo	Toca 30 segundos, desvia para
Horário Saída de Chamadas	Favor Selecionar
Identificação Chamada (callerid)	
Interface	Favor Selecionar porta: Favor Selecionar
Provisionamento - Configuração automática dos telefones	
Template Provisionamento	KHOMP IPS-200
MAC	00:1B:44:11:

* - Required field

Salvar Limpar Voltar à Lista >>>

5.3.6. Dispositivos Móveis (Mobile dispositive)

This screen allows you to configure exclusive access for mobile devices on the Android platform. The required application can be downloaded through the following link: http://ipaddressvoip/android/ipbx.android.apk.

With the Android application, you can use a webservice to request calls for yourself and the destination phone.

In addition to the Android application, an additional manual exclusively for API usage has also been provided. You can find the document in the same structure as the current document is made available.

5.3.7. Provisionamento (Provisioning)

Provisioning is used to configure a device, such as a VoIP phone, without needing to access the device's configuration menus.

The process works as follows: when you purchase a device, like a physical phone from GrandStream, the device's MAC address (a unique identifier) is displayed on the device's packaging.

In your company's DHCP settings, you need to add the TFTP (Trivial File Transfer Protocol) service name available for this type of device.

Here's an example of DHCP configuration in Linux format, considering editing the file /etc/dhcp/dhcpd.conf:

```
option tftp-server-name "tftp_server_address";
```

This way, when the phone is unboxed and connected to the network, it will not only receive an IP address but also all the necessary phone configurations that need to be set up.

In the mentioned phone configuration screen, at the bottom of the page, you should enter the MAC address of your device.

The IPBX system is already prepared for the following models:

- AudioCodes 310HD
- GrandStream GXP1400
- KHOMP IPS-200
- KHOMP IPS-212

However, this screen allows the configuration of any other device that supports provisioning in the standard format.

5.3.8. Painel de Operações (Operation Panel)

This is the available configuration to customize FOP2, as explained earlier, allowing each user to have access only to specific queues or extensions.

5.3.9. Desvios Programados (Desviacion group)

You can configure a diversion group based on time and holidays. To do this, you simply specify the days of the week and the hours during which the diversion of the extension configured for this group should take place.

Additionally, you can set a specific behavior for diversion on a particular date or holiday.

Example:

Dia da Semana	Hora Início 01	Hora Fim 01	Desvia para	Hora Início 02	Hora Fim 02	Desvia para	Hora Início 03	Hora Fim 03	Desvia para	Hora Inicio 04	Hora Fim 04	Desvia para
Segunda-Feira	08:00:00	12:00:00		13:00:00	17:48:00							
Terca-Feira	08:00:00	12:00:00		13:00:00	17:48:00							
Quarta-Feira	08:00:00	12:00:00		13:00:00	17:48:00							
Quinta-Feira	08:00:00	12:00:00		13:00:00	17:48:00							
Sexta-Feira	08:00:00	12:00:00		13:00:00	17:48:00							
Sabado	08:00:00	12:00:00		13:00:00	17:48:00							
Domingo	08:00:00	12:00:00		13:00:00	17:48:00							
Data Especifica/Feriado												

There is also a dedicated screen for holidays or special dates, where you can add descriptions. Once updated, it will follow the same configuration as the screen mentioned earlier.

5.4. Agenda (Appointments)

In the agenda field, it is possible to configure corporate phones that are preconfigured and can be dialed through shortcuts using the *7 screen followed by the shortcut number of the corporation. It is possible to provide cost center rules here, ensuring that everyone using the shortcut is using the specific cost center for the call.

Additionally, it is possible to configure a company blacklist, where phone numbers attempting to sell items or for any other reason requiring blocking can be blocked. The blocks can be applied immediately or through a password validation provided during the call. The passwords are randomly generated and communicated orally.

The callback function is also available in the scheduling menu. If a specific callback-type extension receives a call, the call is disconnected and immediately returned to the originating number using a specific interface. This allows, for example, having a corporate plan for the company, and when necessary, someone who is outside the office can call the callback number, and the call is automatically returned by the company using one of the mobile phones connected to the system.

5.5. Contratos (Contracts)

The configuration includes the possibility to define 3 different types of contracts for accounting VoIP calls, with more details available on the screen. Telephone contracts are agreements between a telephone service provider and a customer, establishing the terms and conditions for the use of telephone services.

Telephone contracts can vary in terms of billing and rates, depending on the customer's needs and requirements. Some common elements in telephone contracts include:

- Rate plans: Contracts usually establish the rate plans applicable to different types of calls, such as local, national, or international calls. These plans may include fixed rates per minute, discounted rates during specific hours, or packages with included minutes.

- Cost per increment: Call increment refers to the duration of a call. Contracts may specify the cost per increment, i.e., the amount charged for each time interval (e.g., per second, per minute) during a call.
- Initial increment: The initial increment is the minimum time interval for which a call is charged. For example, if the initial increment is set to 30 seconds, any call shorter than that limit may be rounded up to 30 seconds and charged accordingly.
- Limits and restrictions: Contracts may also include usage limits, such as the maximum number of minutes or calls allowed within a certain period. Additionally, restrictions may be applied to certain types of calls, such as calls to premium numbers or value-added services.
- Accounting tools: Contracts can specify the tools and methods used for call accounting, such as call logging systems, billing reports, and auditing processes.

It is important to configure these types of contracts correctly to ensure accurate accounting of VoIP calls and enable efficient management of telephony costs within the organization.

5.6. Relatórios (Reports)

It is possible to list the made and received calls and also track this information through graphs. For example, a pie chart can be used to identify the cost center that spent the most on calls during a specific month. Additionally, reports can be obtained on the cost of each department, the percentage of usage of contracted interfaces, call diversion reports, and various other types of reports.

Some of these reports also allow for the download of call recordings made by the extensions. This enables the review and subsequent analysis of these recordings if necessary.

These reporting and tracking functionalities provide a comprehensive view of VoIP call usage, allowing the company to assess performance, control costs, and make data-driven decisions regarding telephone communications.

5.7. Grupo de Captura (Capture group)

The system allows for the configuration of the call capture group.

The call capture group is an association of extensions that enables the capturing of calls from other extensions that are ringing. This way, extensions belonging to the same group can answer the calls by pressing the *0 key combination on their phones. This facilitates collaboration among users and allows ongoing calls to be answered by any member of the call capture group.

5.8. Gravações (Records)

In the telephone system settings, it is possible to define the duration of call and conversation recordings. Additionally, you can specify the location where the recordings will be stored.

It is also possible to access the recordings directly on the server. The recordings are done in the GSM format, using the GSM codec. On average, each minute of recording consumes around 92 Kb of disk space.

The GSM format is an audio codec widely used in telephony, known for its compression efficiency. It allows for the recording of calls in a compact format, saving storage space without significantly compromising audio quality.

5.9. Centro de Custo (Cost Center)

A cost center (Centro de custo) is a record used to categorize and separate costs into specific areas within a company. This separation is done to allow for a more detailed accounting of the organization's costs and expenses.

Each area or department of the company can be designated as a cost center. This is especially useful when different areas have separate budgets or when it is necessary to closely monitor the costs of each department.

By registering cost centers, it is possible to allocate specific expenses and costs to each area of the company. This provides a clearer view of the expenses associated with each department, enabling a more accurate analysis of the financial performance of each area and facilitating strategic decision-making.

By using cost centers, companies can have more effective control over expenses and a better understanding of how financial resources are being allocated across different areas. This assists in financial management and strategic planning, helping to identify areas of greater efficiency or potential cost reduction opportunities.

5.10. Horários (Hour Registration)

This is the registration of the time slots during which you authorize outgoing calls from the company. The control is done per extension, and the registered list appears in the "Call Out Time" field.

5.11. Call Center

A telephony call center is a dedicated phone service unit that specializes in providing support and services related to the field of telephony. It is a department or company that manages a large volume of customer calls, offering technical support, customer service, sales, and other telephony-related services.

Telephony call centers are responsible for handling a variety of customer inquiries and requests, such as product information, technical assistance, issue resolution, service activation, cancellations, payments, and complaints. They can serve both individual customers and businesses, depending on the type of service provided.

These phone service units are typically equipped with advanced telephony systems and call management software to ensure that calls are properly routed and efficiently handled. Call center agents are trained to handle different types of calls and provide quality service to customers.

In addition, telephony call centers may also be involved in telemarketing activities, making outbound calls to offer products, services, or conduct market research.

In summary, a telephony call center is a specialized service center that handles telephone calls related to telephony services, providing support, customer service, and other solutions to meet customer needs in this field.

5.11.1. Filas (Queue)

Call queues are mechanisms used to organize and manage the order of customer service, aiming to ensure fair and efficient service. The main objective is to minimize wait times and provide a positive customer experience during the service process.

When setting up a call queue in the IPBX system, there are various options and configurations available to customize the queue's operation. Some of these configurations include:

- Queue name(Nome da fila): It is possible to assign a name to the queue to facilitate organization and identification in the settings menu.
- Call distribution strategy (Estratégia de chamadas DAC): This strategy defines how calls will be distributed among available agents. Strategies such as random, round-robin (sequential), assigning to agents with the least workload, agents who have been least recently called, or to all agents simultaneously can be used.
- Ring time (Tempo de toque): It sets the time in seconds that a call will ring at each agent before being redirected to the next one.
- Service hours (Horário de atendimento): It is possible to define the start and end time of the queue's service. This allows calls to be directed correctly according to the established schedule.
- Maximum user waiting time (Tempo máximo de usuário em fila): Determines the maximum time a user can remain on hold in the queue. If this maximum time is reached, diversion settings can be configured, such as call redirection or termination of the call.
- Recording configuration (Configuração de gravação): Allows defining whether calls made in the queue will be recorded or not.
- Service hours and additional information audio (Áudios de horário de atendimento e informações adicionais): It is possible to upload audio files to inform users about the queue's service hours and provide other relevant information before the service.

- Strategies for logging out agents (Estratégias para deslogar atendentes): Allows configuring automatic actions to log out agents if the softphone is not registered, after the configured service hours, or when the last user in the queue is serviced, even if it is after the service hours.
- Maximum number of users in a queue (Número máximo de usuários em uma fila): Defines the maximum number of users that can wait in the call queue simultaneously, preventing the queue from becoming excessively long.
- Time between calls (Tempo entre atendimentos): Determines the time, in seconds, that an agent has between one call and the next.
- SLA (Service Level Agreement): It is possible to record information related to SLA, such as wait time targets, average handling time, and other performance indicators.
- Additionally, it is important to mention that it is possible to assign a manager responsible for monitoring the queue's service. These management features will be explained later on a specific screen.

These configurations allow customization of the call queue's operation according to the company's needs, ensuring efficient management and meeting the demands of customer service.

5.11.2. Agentes (Agents)

A priority configuration is related to the hierarchy of service in call queues. When configuring agents or attendants, it is possible to specify which queues they can log in to and handle calls.

Furthermore, it is possible to define an agent's priority in relation to the queues they are enabled for. This means that an agent can have a higher priority in a specific queue compared to another queue.

This priority determines the order of call handling when multiple agents are available. For example, if an agent is considered more prioritized in one queue compared to another, they will be called upon to handle calls from that queue before less prioritized agents.

This configuration is useful when there is a need to ensure that certain agents handle calls in specific queues before others. For example, it may be necessary to ensure that a specialized agent in a particular subject or with more experience is the first to handle calls in a specific queue.

By configuring priorities, it allows for greater flexibility and control over call distribution and the order of handling in different call queues. This ensures better management and directs calls to the most appropriate agents for each situation.

5.11.3. Cadastro de pausa (Break Registration)

The break registration feature allows agents to record the moments when they need to temporarily interrupt their activities in the call center. These breaks can be used for various purposes, such as going to the restroom, performing back-office tasks, or carrying out other activities necessary for subsequent handling of calls.

The main objective of break registration is to provide information for the reports on the agent dashboard, which will be explained later. When an agent selects a break and indicates the reason, both the manager and the call center are informed about the reason for the break. This ensures that the call center does not route any more calls to the agent on break, avoiding unnecessary interruptions during that period.

Additionally, it is possible to specify the number of minutes an agent will work per day. This allows the call center to control and record the hours worked by the agents, providing important data for productivity calculation and resource management.

This functionality enables better monitoring of the agents' working time, recording of breaks taken, and optimization of call routing, taking into account the moments when agents are available for handling calls. This contributes to more efficient call center management and monitoring of agent productivity throughout the day.

5.11.4. Horário de atendimento (Queue Service Hour)

Register the service hours during which the queue will receive calls for distribution.

5.11.5. Pesquisas (Survey)

The survey registration feature allows you to create customized surveys to be conducted at the end of calls in the telephone system. By registering a survey, you can transfer the call to a specific extension designated as "survey" or configure an extension to always be of the survey type.

In the survey registration, you can enter the questions you want to ask users. Additionally, you can upload audio files containing the questions, which can make the interaction more intuitive and user-friendly.

You also have the option to register valid answers for each question. This allows the answers to be recorded, and later you can generate comprehensive reports of your surveys, analyzing the obtained responses.

This functionality is useful for collecting customer feedback, conducting satisfaction surveys, gathering information about the quality of the service provided, or any other type of survey that is relevant to your business.

With the survey reports, you will gain valuable insights into customer perception, enabling you to identify areas for improvement, evaluate the effectiveness of the service, and make strategic decisions based on the results obtained.

In summary, the survey registration feature offers the possibility to create and conduct customized surveys, obtain user responses, and generate comprehensive reports for analysis and decision-making.

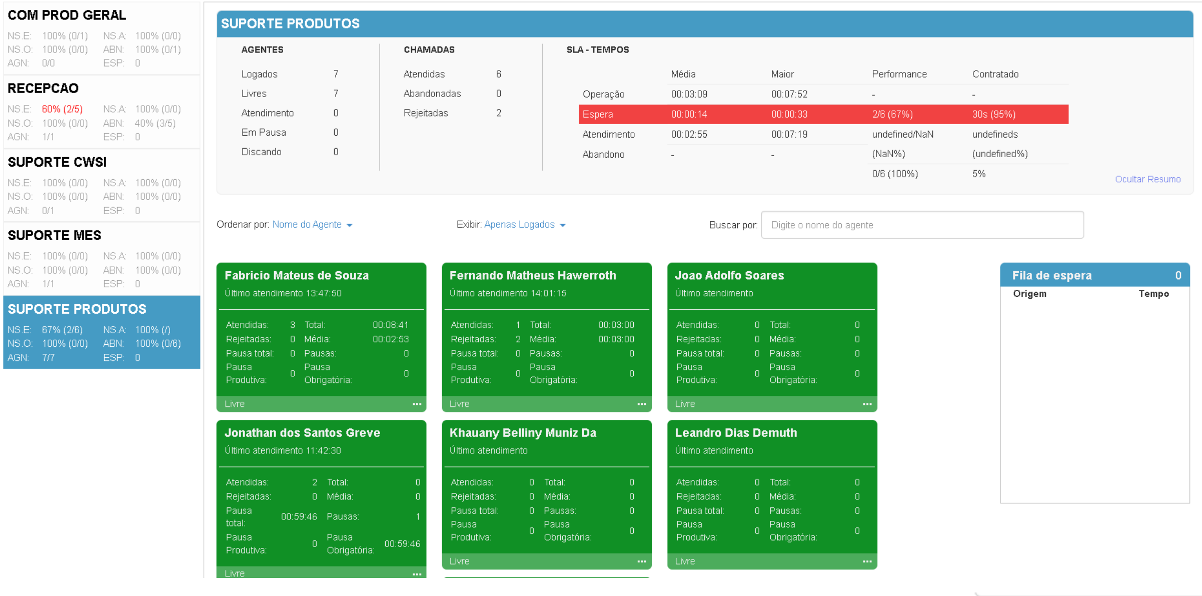
5.11.6. Monitor Painel de agentes e painel de gestor (Dashboard Agents and Manager)

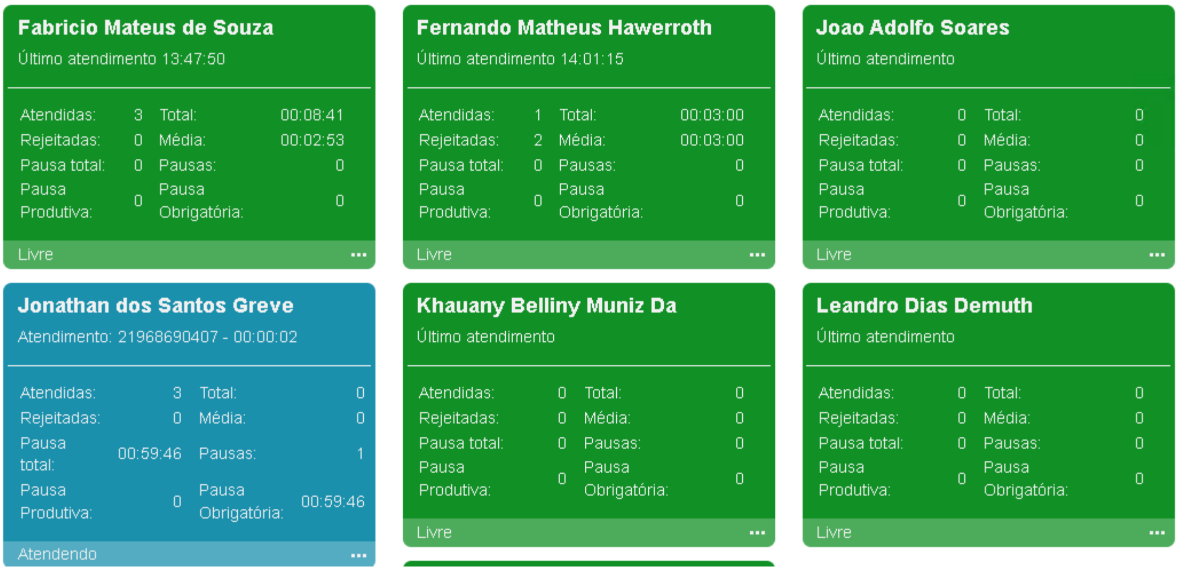
The manager's screen is a highly interactive and visually rich tool that allows the call center manager to have a complete and real-time overview of the state of the service. Some of the information that can be viewed on this screen includes:

- Logged-in agents (Atendentes logados): It shows which agents are currently logged into the system and ready to handle calls.
- Agents off-duty (Atendentes fora do atendimento): It displays the agents who are temporarily unavailable to handle calls, whether they are on a break, engaged in other tasks, or for any other reason.
- Incoming callers (Clientes ligando): It identifies how many customers are currently calling the call center, waiting to be attended to.
- Average handling time (Tempo médio de atendimento): This metric shows the average time agents are taking to complete each call.
- Average wait time (Tempo médio de espera): It displays the average time customers are waiting in the queue before being answered.
- Abandoned calls (Chamadas abandonadas): It shows the number of calls that were abandoned by customers before being answered by an agent.
- Ongoing calls (Ligações em curso): It indicates the number of calls that are currently in progress, meaning they are being handled by an agent.
- Log out agents (Deslogar atendentes): It allows the manager to log out agents from the system, if necessary.
- Calls answered (Ligações atendidas): It shows the total number of calls that have been successfully answered by agents.
- Information on whether the agent has worked enough time during their shift in the service.
- Calls rejected by agents (ligações rejeitadas): It indicates the number of calls that were rejected by agents, whether due to being busy, unavailable, or for any other reason.

These pieces of information and metrics enable the manager to have a comprehensive view of the call flow and the performance of the service team. Based on this information, the manager can make decisions and necessary adjustments to optimize the service, reduce customer wait time, and improve the quality of the service provided.

Example:





In the agent's screen, several functionalities are provided to assist in the customer service work. Some of the options that the agent may find on this screen are:

- Login and logout from the service queue: The agent can indicate their availability to receive calls by logging into the service queue when ready to handle calls and logging out when unavailable.
- View recent calls: It is possible to view a record of the agent's recent incoming or outgoing calls. This allows the agent to have a history of interactions with customers and facilitates the tracking of previous conversations.
- Listen to recent calls: The agent has the ability to listen to recordings of the most recent calls that have been recorded. This can be useful for reviewing the content of conversations, obtaining additional information, or ensuring the quality of service.
- Take a break from handling calls: The agent can select a predefined break option to inform that they need to temporarily step away, whether it's for a restroom break, performing a background task, or any other reason. This break will be recorded for

reporting purposes, and the system will stop forwarding calls to the agent during this period.

- Ability to see when the time limit for the agent's call has been reached.
- Make calls through the interface: The agent can make calls to external numbers using the system's own interface, without the need for an external telephone. This facilitates the process of making calls related to customer service directly from the agent's screen.

These are just some of the functionalities that may be available on the agent's screen. The goal is to provide the necessary tools for the agent to efficiently perform their customer service activities and have access to relevant information to improve customer interactions.

5.11.7. Relatórios de call center (Call Center reports)

In the call center, various visualizations and reports are available to monitor and analyze the performance of the service. Some of the available visualization and reporting options include:

- Average time per agent: It is possible to visualize the average handling time for each agent, allowing for the evaluation of individual efficiency and productivity within the team.
- Charts of call volume over time: These charts show the distribution of call volume over time. They help identify demand patterns and variations in workload throughout the day, week, or month.
- Charts of call distribution by queue: These charts display the distribution of calls among different queues. This allows identification of overloaded or high-volume queues.
- Queue productivity report: This report provides information on queue performance, including average wait time, average handling time, abandoned call rate, and other relevant metrics. It assists in evaluating queue efficiency and identifying potential improvements in the service process.
- Reports with call details and recordings: These reports present details about the calls, such as time, call duration, customer number, responsible agent, and other relevant data. Additionally, these reports may include links to access call recordings, enabling a detailed review of customer interactions.

These are just a few of the visualization and reporting options available in the call center. The aim is to provide accurate and detailed information about service performance, enabling comprehensive analysis and assisting in decision-making to enhance the quality and efficiency of the service provided.

5.12. Salas de conferência (Conference Room)

The call center provides settings for fixed conference rooms, enabling voice calls and video conferences. Some of the available configuration options for these rooms include:

- Password access control: It is possible to set a password for each conference room, ensuring that only authorized participants can join the call. This helps maintain the security and privacy of the conferences.
- Conference expiration date: It is possible to set an expiration date for conference rooms, after which they will no longer be available for use. This allows for more precise control over the duration of conferences and prevents misuse or prolonged use of the rooms.

In addition to these basic configurations, there are other advanced options that may be available in the call center, such as:

- Participant limit: It is possible to set a maximum limit of participants for each conference room, ensuring that there are no excessive users in the call.
- Conference recording: It is possible to configure the recording of conferences held in the fixed rooms. This allows calls to be recorded and played back later for review or reference.
- Screen sharing: Some call centers offer the option of screen sharing during conferences, allowing participants to share documents, presentations, or other visual content.

These fixed conference room settings provide flexibility and control over voice calls and video conferences conducted in the call center. The additional features, such as access control, expiration date, participant limit, and recording, contribute to a secure and efficient conference experience.

5.13. Fax

To receive and send faxes in the call center, you can use the T.30 technology, which is the default standard. When using this type of fax, it is necessary to use codecs such as alaw/ulaw. However, if you want to adopt the T.38 standard, you need to configure the Asterisk fax, as previously mentioned in the document.

The T.30 protocol is widely used for fax communication in telephony systems. It defines the standards and procedures for transmitting fax data through traditional telephone networks. With T.30, it is possible to send and receive faxes in image format.

On the other hand, the T.38 protocol is a fax protocol that allows fax transmission over IP networks, such as the internet. It converts the analog fax signal into digital data for transmission over the IP network. T.38 is especially useful in VoIP environments where traditional fax transmission can be problematic due to the nature of IP packets.

By configuring the Asterisk fax to use the T.38 standard, you enable fax transmission over IP networks, which can offer advantages in terms of transmission quality and reliability. This configuration should be done following the instructions provided at the beginning of the document or according to the specific guidelines of the system you are using.

Therefore, when configuring the fax in the call center, you can choose to use the standard T.30 technology or configure it to use the T.38 protocol, depending on your fax transmission needs and requirements.

5.14. Gestão SMS

The creation of groups for sending and receiving SMS in the call center is possible when using a Khomp GSM interface, along with the contracted cards with the mobile operator for SMS sending. This configuration allows you to create specific groups and efficiently direct text messages to these groups.

Firstly, it is necessary to have a Khomp GSM interface installed and configured in the call center. This interface enables communication between the call center and the GSM network of mobile operators. It is important that you have the necessary contracts with the operator for SMS sending, as sending text messages requires a partnership with the operator for proper data transmission.

After installing the interface and configuring the operator's cards, you can proceed with the creation of SMS groups. The groups can be configured according to your specific needs and criteria. For example, you can create groups for different departments in the company, customer groups, or any other segmentation that is relevant for sending and receiving text messages.

Once the groups are configured, you can use the call center to send and receive text messages to these groups. This allows for agile and efficient communication with the members of the selected groups.

It is important to note that sending and receiving SMS through the Khomp GSM interface is subject to the policies and limitations of mobile operators. Therefore, it is necessary to ensure compliance with regulations and agreements established with the operator to avoid any issues or restrictions in SMS sending.

In summary, the creation of groups for sending and receiving SMS in the call center requires the use of a Khomp GSM interface and the necessary contracts with the operator for SMS sending. This configuration allows for efficient communication through text messages directed to the defined groups, providing a more agile and personalized service experience.

5.15. Ura reversa (Reverse IVR)

In the call center, there is the possibility to make calls to customers and offer an Interactive Voice Response (IVR) system with interactive options. An IVR, also known as an Interactive Voice Response, is an automated system that provides a menu of options to customers through voice recordings.

To use this functionality, you need to set up the IVR information in the call center. This includes recording the voice messages that will be played to customers and defining the available options in the menu.

When calling a customer, the IVR will be activated, and the customer will hear a recording presenting the available options. For example, the menu may prompt the customer to press certain numbers to select options like "Press 1 for technical support, press 2 for customer service, press 3 for sales, etc."

Depending on the option selected by the customer, the call can be forwarded to a specific extension, a call queue, or other destinations configured in the call center. This allows directing customer calls according to their needs and preferences.

The use of IVR offers several advantages, such as automating initial customer interactions, streamlining call screening and routing, and providing a more personalized experience for customers.

It is important to note that configuring the IVR in the call center requires knowledge and proper configuration of the available options and call destinations. It is recommended to have specialized professionals or follow the guidance and documentation provided by the call center to correctly set up the IVR and ensure its proper functioning.

In summary, the possibility to call customers and offer an IVR system with options is available in the call center. To use this functionality, you need to set up the IVR information, including voice recordings and menu options. This allows directing customer calls based on their choices, providing a more efficient and personalized experience.

5.16. Configurações

5.16.1. Controle de acesso e Área Admin (Access Control and Admin Area)

To control access to the screens in the call center in a granular manner, there are two available options: granting access to screens individually through the administrative area or configuring access groups in the access control area.

In the administrative area, a user with administrator privileges can define which screens will be accessible for each user or user group. This means that it is possible to specify which screens will be available and which will be restricted for each user profile.

In the access control area, it is possible to configure access groups with predefined permissions. Access groups allow grouping users with the same permissions and assigning these groups to the call center screens. This way, access can be controlled in a more simplified manner by defining permissions at the group level.

To access the administrative area, the user "admin" must enter the password "_cell" in the corresponding field, located in the upper left corner next to the IPBX central menu.

By using these options, it is possible to precisely control which screens and resources of the call center each user or user group will have access to. This helps ensure the security and privacy of information, allowing for more efficient control over the available functionalities for each user profile.

5.16.2. Auditoria (Audit)

The call center has an auditing feature that records all changes made in the system. This auditing is a kind of activity log where every modification made in the call center configuration is registered and stored for control and historical purposes.

By accessing the auditing reports, it is possible to view detailed information about the changes made, including the date and time, responsible user, type of modification, and specific details of the alteration performed.

This auditing feature is useful for tracking and monitoring changes made in the call center, providing greater transparency and control over the configurations made. It also helps identify any errors or issues, allowing for quick and precise correction.

By utilizing the auditing reports, it is possible to have a comprehensive view of the change history, facilitating management and monitoring of modifications made in the call center. This contributes to the security and smooth operation of the system by maintaining a detailed record of activities performed by authorized users.

5.16.3. Backup/Restore

The telephone system offers the possibility to perform manual backups, allowing you to create a backup copy of the current data and configurations of the system. This functionality is crucial before making critical or significant changes to the system's configuration because in case of any issues or unforeseen events, you can restore the system to its previous state using the backup.

When performing a manual backup, various essential elements of the telephone system are saved, such as extension configurations, call queues, pickup groups, IVR (Interactive Voice Response) settings, audit logs, among others. The backup can be stored in a secure location, such as an external server, external storage device, or in the cloud, ensuring that the data is protected and available for eventual restoration.

Performing manual backups is a recommended practice to ensure the integrity and availability of the telephone system's data. This way, you can have peace of mind knowing that you can recover the system if necessary, minimizing potential impacts caused by failures or errors during the changes made.

5.16.4. LDAP

Integration with LDAP (Lightweight Directory Access Protocol) allows the IPBX telephone system to integrate with an LDAP server, whether it is Windows-based or Linux-based (such as Samba), to retrieve user data and associate it with the telephone extensions in the system.

When configuring LDAP integration, you will need to provide the correct information for the system to connect to the LDAP server, such as the IP address or server name, access credentials, and other relevant parameters. These details may vary depending on the LDAP server being used.

Once the configuration is correctly set up and the connection to the LDAP server is established, the IPBX system will be able to retrieve user data directly from the LDAP server. This means that users registered in the LDAP server will be recognized by the telephone system.

In the configuration interface, you will be able to view the users imported from the LDAP server. From there, you can associate these users with the telephone extensions in the system, ensuring that each extension is identified with the correct user.

Integration with LDAP brings benefits such as centralized user management and automatic synchronization of user information with the telephone system. This means that any updates or changes made in the LDAP server will be reflected in the configuration of the telephone system.

It is important to follow the specific guidelines provided by your system and LDAP server when performing the integration with the IPBX telephone system, ensuring proper configuration and compatibility between the systems.

5.16.5. Parâmetros do sistema (System Parameters)

When setting up the telephone system, one of the first steps is to define the range of extensions that will be served by the system. This involves determining the extension numbers that will be assigned to users and devices connected to the telephone system.

In addition, there are other customizable parameters that are global to the telephone system. Here are some examples of these parameters:

- Default message: You can configure a default message that will be played by the telephone system when a customer contacts it. This message can include information about the company, business hours, menu options, etc.
- Maximum call duration: You can set a maximum duration for calls made through the telephone system. This can be useful to limit the duration of each call and ensure better resource management within the system.
- Webservice password: The telephone system may offer a webservice for integration with other systems or applications. In such cases, it is important to configure an access password for the webservice to ensure security and proper authentication of access.

- SMTP settings: For email notifications, the telephone system may require the configuration of SMTP server parameters, such as address, port, authentication, encryption, and others. These settings are important for the proper functioning of the email features in the system.
- Management and agent screen refresh time: You can define the refresh time for management and agent screens in the telephone system. This determines how often the information is updated on the screens, providing a real-time view of call status and agent availability.
- Carrier/operator number: It is important to configure the carrier/operator phone number in the telephone system. This number will be used to forward calls that do not match any menu options or in case of failures in the call routing.

These are just a few examples of global parameters that can be configured in a telephone system. Each system may have its own options and specific configurations. It is important to review the documentation or configuration guide provided by the manufacturer or provider of the telephone system to obtain detailed information about the available parameters and options.

5.16.6. Sincronismo (standby)

Yes, it is possible to configure a second standby call center to function as a backup. This configuration is known as a redundancy or failover system.

In this scenario, the second call center remains in standby mode and is activated if there is a problem with the primary center. The checking frequency can be configured to occur every 5 minutes, for example.

When a problem is detected in the primary center, you can access the standby center and use the "Promover para servidor de produção" option to activate it as the primary server. Thus, it takes on the call handling and management functions.

In addition, there are other available configurations, such as the synchronization speed between the centers. This determines how quickly information is updated and synchronized between the servers.

Regarding recordings, you can configure whether they should be automatically transferred to the standby center or remain only in the primary center. This configuration depends on the company's needs and policies regarding call recording storage and management.

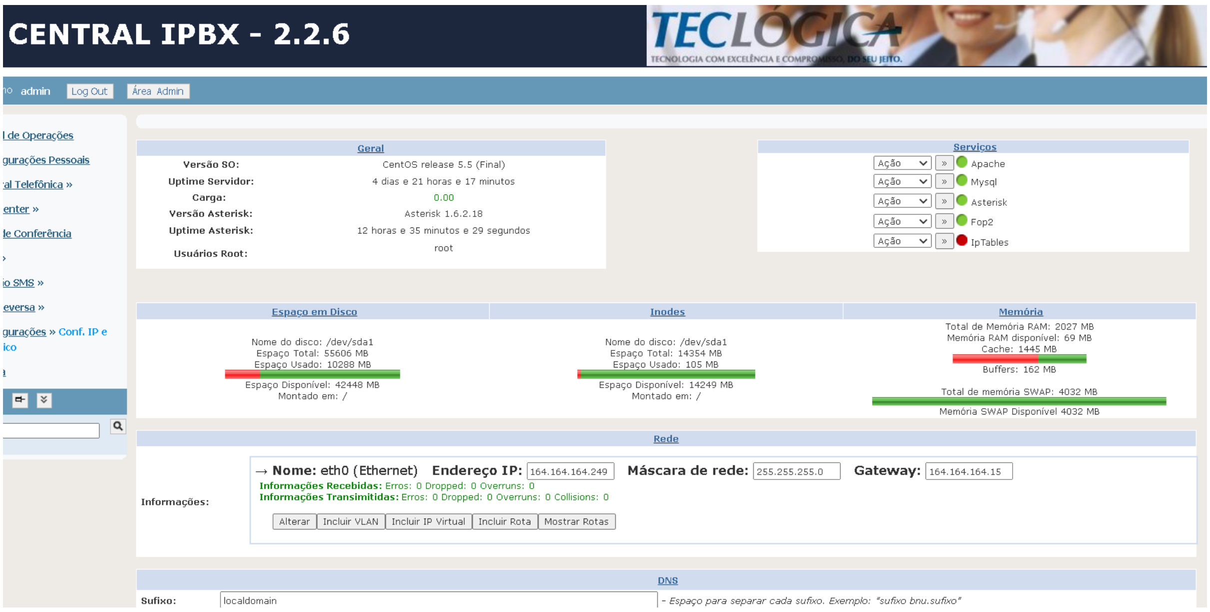
It is important to note that implementing a redundancy or failover system requires proper configuration of the equipment and software, as well as ensuring that both servers are in proper operating condition. It is recommended to consult the documentation or technical support of the call center manufacturer for detailed guidance on how to configure and use the redundancy feature in your specific call center.

5.16.7. Configurações IP/Diagnósticos e serviços (IP Settings/Diagnostics and Services)

The mentioned screen displays detailed information about the overall status of the machine on which the solution is running. This information may include:

- IP Configurations: Shows the configured IP address for the server, both the local IP address and the public IP address if applicable. This allows verifying the network settings and ensuring that the server is properly connected to the network.
- Services: Displays the status of services or processes running on the server. This may include specific services of the telephone central solution, such as call management service, integration service with other systems, among others. This information is useful to check if the services are running correctly.
- Inode Count: Inodes are data structures used by the file system to store information about files and directories. The available inode count is important to ensure that the server has sufficient capacity to store files and directories.
- Disk Usage: This information shows the total storage capacity of the disk used by the server and the amount of available space. Monitoring disk space usage is important to ensure that there is enough space for the operations of the telephone central, such as storing call recordings, logs, and other data.
- Memory: Displays information about the server's memory usage, including the total amount of available memory, the amount of memory in use, and the amount of free memory. This helps monitor the server's resource usage and ensure that there is sufficient memory for the smooth operation of the telephone central and other running services.

These are just some of the pieces of information that can be displayed on this general server status screen. Depending on the specific telephone central solution used, other relevant information may be provided to effectively monitor and manage the server.



5.16.8. Tarifações locais (local rates)

To implement differentiated pricing for local area codes (DDDs) in a telephone system, the following configurations are typically required:

- Define local DDD codes: Identify the DDD codes corresponding to the local calls you want to tariff differently. For example, if your company is located in São Paulo (DDD 11) and you want to apply a special rate for calls within the same area code, you need to configure the DDD code 11 as local.
- Configure tariff plans: Most telephone systems allow you to create different tariff plans to apply call charges. You will need to configure a specific tariff plan for local calls with differentiated rates. In this plan, you can define the rates, per-minute charges, or any other applicable cost calculation method.
- Associate extensions with the tariff plan: To ensure that calls made from extensions are correctly tariffed, you must associate each extension with the corresponding tariff plan. This association can be done through settings in the telephone system, where you indicate which tariff plan will be used for each extension.
- Configure call routing rules: It is possible to configure specific call routing rules to direct local calls to the differentiated tariffing. For example, you can create a rule that directs all calls destined for DDD 11 to the local call tariff plan.

It is important to note that the configurations may vary depending on the specific telephone system solution used. It is recommended to consult the documentation or support of the telephone system provider for specific guidance on how to configure differentiated pricing for local area codes.

5.17. Ajuda

The quick help menu provides information on shortcuts and concise instructions for efficiently using the system. It is a handy tool that offers users easy access to information about system features and usage.

5.18. Self-service programming (flowchart)

To create an auto-attendant menu, mechanisms are used that read the tables listed below. However, there is no specific screen for inserting or uploading audio for the menu. If these actions need to be performed, you will need to place the audio files in a directory and manually insert them into the mentioned tables:

- Table "ipbx_fluxograma"
- Table "ipbx_fluxograma_adapters"
- Table "ipbx_fluxograma_dados_executor"
- Table "ipbx_fluxograma_decisor"
- Table "ipbx_fluxograma_desligar"
- Table "ipbx_fluxograma_desviador"
- Table "ipbx_fluxograma_executor"
- Table "ipbx_fluxograma_executor_variaveis"

- Table "ipbx_fluxograma_gravador"
- Table "ipbx_fluxograma_iniciador"
- Table "ipbx_fluxograma_mensagem"
- Table "ipbx_fluxograma_questionador"
- Table "ipbx_fluxograma_variavel"
- Table "ipbx_fluxograma_vocalizador"

These tables contain essential information for configuring the auto-attendant menu and should be manually updated if you need to add new audio or make adjustments to the menu options.

6. Recommendations next steps (Roadmap):

It is crucial to update the operating system, Asterisk, and related services for the overall solution. Since the project has been inactive for a few years, there may be security issues that can affect the company currently using the solution. To ensure system security, it is recommended to:

- Perform a vulnerability analysis on the web system using tools like OWASP or other vulnerability analysis tools.
- Enhance the login screen to prevent brute force attacks.
- Upgrade the version of jQuery to a more recent one, reducing the possibility of security issues.

Furthermore, it is important to develop interfaces for the self-service functionality as the Asterisk code is already prepared to receive this interface layer.

Creating additional graphical screens to better understand the call status is recommended. The existing API can be utilized for this purpose.

Other necessary improvements include:

- New call tracking report: Provide details about each step taken by calls within the phone system until completion.
- Implement a new event visualization screen: Facilitate the identification of issues in the phone system.
- Add call information functionality for Android/iPhone devices: Provide information about dialed and received calls, and make the application available on Google Play and the Apple Store.
- Add the option of trunk dialing using (Utech) portability or others.
- Create detailed reports for calls by queue and agent group, queue history by group and date, queue history by group and hour, agent details by queue group, abandoned call details by group, and call details by group (Antonio Oliveira).
- Allow the use of MCDU with 4 or 8 digits to handle overlap of ranges (last 4 digits).
- Create a new report to measure agent work time (Agent Availability Report).
- Add an alarm clock function for hotel use.
- Modify the user interface to support other languages.