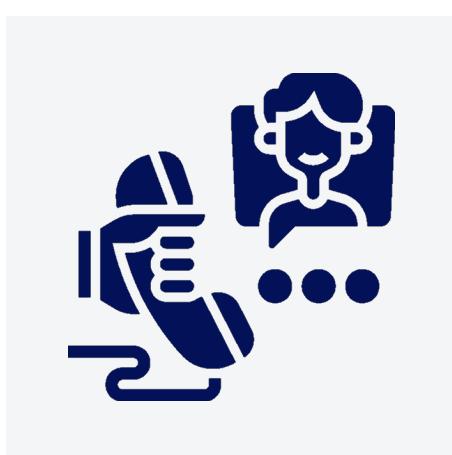




Key features



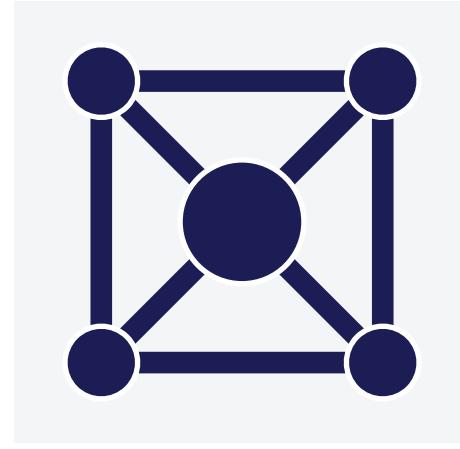
Interactive Voice Response (IVR)

Difuse PBX takes customer interaction to new heights with its sophisticated IVR capabilities. Guide callers with customized voice menus, automate routine inquiries, and route calls efficiently, ensuring optimal customer engagement and reduced wait times.



IPv6 Support

Difuse PBX embraces the future with comprehensive IPv6 support. As the Internet evolves, Difuse ensures your business stays connected, leveraging the expanded addressing capabilities and enhanced security of IPv6.



Expanding Connectivity

Difuse PBX stands as a communication lynchpin. Integrate analogue lines, PRI interfaces, and GSM gateways, ensuring businesses stay connected across diverse communication channels, be it traditional telephony or mobile networks.



Seamless Integration

Difuse PBX champions interoperability. Integrate seamlessly with SIP-compliant PBX systems from various brands, ensuring a cohesive and flexible communication ecosystem for businesses.



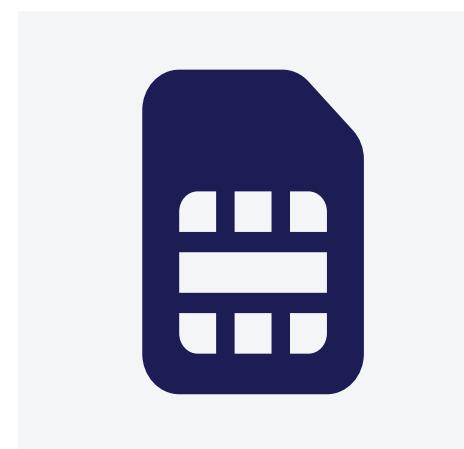
Secure by design

Difuse PBX prioritizes security. Utilizing TLS, it ensures encrypted signaling, safeguarding the initiation and termination of calls. With SRTP, voice data remains confidential and tamper-proof, ensuring eavesdrop-free and secure voice communication.



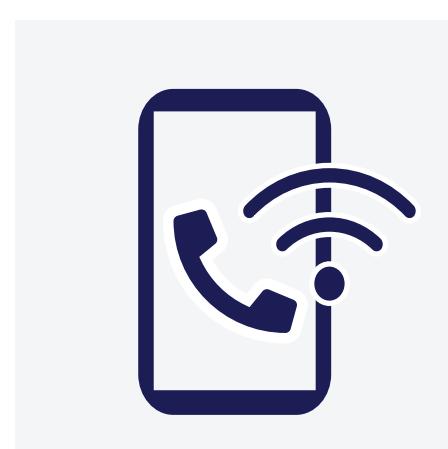
Firewall with Geo IP

Difuse PBX goes beyond standard protection measures. The Geo IP access control further elevates security by allowing businesses to restrict or grant access based on geographic locations. This combination ensures both internal and external communication remains safeguarded.



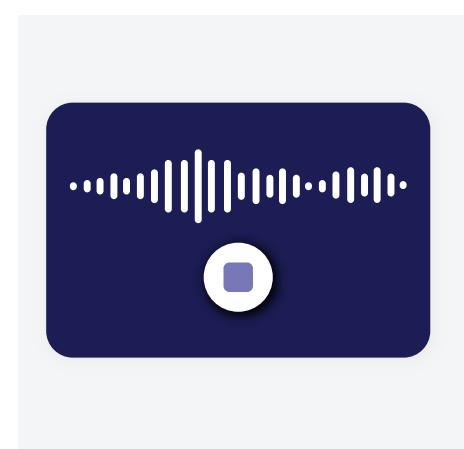
Built in GSM Module

Difuse PBX offers an integrated GSM module, empowering businesses to make and receive calls via GSM networks. Whether it's for cost-saving, ensuring connectivity in areas with poor IP infrastructure, or serving as a backup communication channel, Difuse's GSM capabilities ensure you're always in touch, no matter what.



SIP Compatibility

Difuse PBX prides itself on its versatility, supporting any SIP compliant soft and hard phones. Whether you prefer a specific brand of desk phone or a particular softphone application on your mobile or computer, as long as it adheres to SIP standards, it will seamlessly integrate with Difuse PBX.



Call Recording

Difuse PBX's call recording feature stands as a pillar for quality assurance, compliance, and training. Businesses can effortlessly record interactions, allowing for playback at convenience. This not only aids in dispute resolution but also serves as invaluable material for training and continuous improvement. With easy storage and retrieval mechanisms, managing these recordings becomes a hassle-free task.



Web based Operator Console

Difuse PBX introduces a comprehensive web-based operator panel. This intuitive interface allows administrators to manage call traffic in real-time, oversee extensions, and adjust various communication settings. From drag-and-drop call transfers to monitoring active calls and queues, the panel is a central hub for efficient call management, all accessible from any web browser.



FEATURES

Call Features

Call forwarding, call waiting, call transfer
Call parking
Call pick-up
Music on Hold
Caller ID and Caller ID blocking
WebRTC voice calling

Interactive Voice Response (IVR):

Multi-level IVRs
Time-based routing

Conferencing

Multi-party conference calls
Advanced Call Routing:
Inbound routing based on pattern
Direct Inward Dialing (DID)

Outbound Dialing using access code or pattern match

Integration and Connectivity

Trunking with VoIP providers
Can be integrated with any SIP compliant PBX
Connectivity to traditional PSTN using analog or digital lines using gateways
API integration for custom applications
PPPoE, Static IP and DHCP connection to WAN
Distributed setup with VPN and
TLS Encrypted trunks
No Licensing
Built in GSM Module for Call / Data

Dialplan Scripting

Direct Asterisk config file manipulation
Security
SIP Secure (SIPS)
SRTP for encrypted media
Fail2ban integration for preventing brute force attacks
Fax Support:
Send and receive faxes (T.38 or via audio)
Reporting and Monitoring:
Call Detail Records (CDR)
Real-time monitoring of calls
Call recording

Unveiling the DiFuse Small PBX System: Open Source Excellence

In today's ever-evolving communication landscape, businesses seek platforms that are both robust and adaptable. Enter the PBX or Private Branch Exchange - an essential tool that handles incoming and outgoing calls akin to a digital switchboard.

The Difuse Small PBX System emerges as an embodiment of this need, combining the strengths of open source principles with the time-tested reliability of Asterisk. While it may wear the moniker 'small', Difuse is an epitome of expansive capabilities and unmatched flexibility, perfect for contemporary businesses that thrive on innovation and adaptability.

Empowerment through Open Source

Difuse stands out in its commitment to open source values. Open source isn't just about accessibility; it's about community-driven improvement, transparency, and the freedom to customize. With the Difuse system, you get the advantage of a platform that's been enriched by global contributions, ensuring it stays updated, secure, and ahead of the curve.

Asterisk at its Core

Harnessing the renowned strength of Asterisk, Difuse offers a foundation that is both stable and feature-rich. Asterisk, celebrated for its resilience and versatility in the PBX domain, forms the beating heart of Difuse, assuring users of its time-tested reliability.

Intuitive for All

The brilliance of Difuse lies in its user-centric design. Whether you're an IT guru or a daily caller, the system is remarkably straightforward, minimizing the learning curve and maximizing efficiency.

Tailored to Your Enterprise's Pulse

Each business has its rhythm and nuances. Difuse respects that, offering scalability that aligns with your enterprise's growth trajectory. Plus, its open-source nature means it can be tweaked and tailored to fit precise needs.

Economical, with a Forward-Thinking Approach

Eschewing the high costs associated with traditional systems, Difuse leverages the power of VoIP (Voice over Internet Protocol) for economical communications, especially vital for long-distance interactions.

Seamless Synchronicity with Tools

Difuse's open-source nature means it plays well with other digital tools. Its adaptability ensures seamless integrations, from CRMs to email platforms and beyond, amplifying operational synergy.

In Summation

The Difuse Small PBX System isn't just a communication tool; it's a testament to the power of open source and the unmatched reliability of Asterisk. If you're on the hunt for a communication system that epitomizes transparency, adaptability, and cutting-edge features, Difuse is your go-to choice.

IP PBX Dashboard

PBX / Interactive Voice Response (IVR)

ROUTING & SERVICES

- Network Settings
- Firewall
- Services
- VPN
- Administration

PIX

- Extensions
- Trunks
- Call Routing
- IVR
- Queues
- Conference Rooms
- SMS
- Operator Panel
- CDR
- Block List
- Settings
- ABOUT
- About

Search: Quick Actions: ...

IVR Sounds

Please upload only MP3 files

Drag & Drop your files or Browse

Save Sound

Uploaded Sounds

Name	Action
india_welcome	[Delete]
toms-diner	[Delete]
Welcome	[Delete]

Page Size: 10 | First | Prev | 1 | Next | Last

PBX / Trunks

PBX / Trunks

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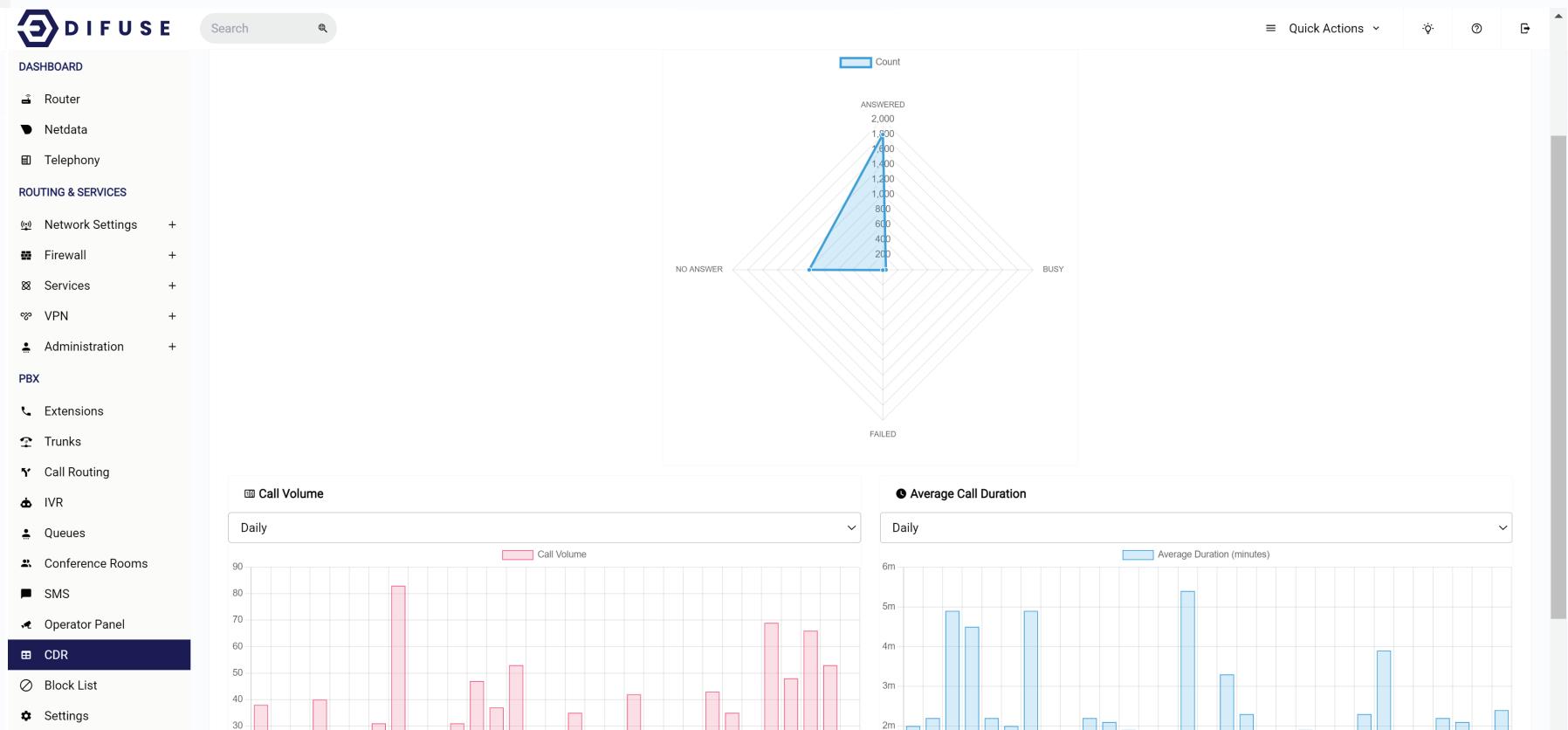
Trunk List

St.	Name	Host	Codecs	Default Action	Created	Action
1	BSNLTrunk	10.191.55.49	G.711 G.722 G.726 G.729	IVR/WelcomeIndia	12:51 PM 07/19/...	[Edit]
2	OfficeDXB	10.254.254.2	G.711 G.722 G.726 G.729	Extension/3000	12:30 PM 08/29/...	[Edit]
3	india	india.pstn.twilio.com	G.711 G.722 G.726 G.729	Extension/3000	7:32 PM 09/18/...	[Edit]

Page Size: 10 | First | Prev | 1 | Next | Last

Create New Trunk | Edit LTE Trunk

Call Detail Report



PBX/Call Routing

PBX / Call Routing

ROUTING & SERVICES

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PIX

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- ABOUT
- About

Search: Quick Actions: ...

Inbound Routes (ID) **Outbound Routes (DO)**

Name	Gateway	Action	Created
0 Rows	No inbound routes found		

Create New Inbound Route

Technical Specification

1 Gigabit Ethernet RJ45 WAN	1
1 Gigabit Ethernet RJ45 LAN	1
SIM Slot (Micro Sim)	1
WAN Modes	DHCP, Static IP & PPPoE
ISDN PRI & Analogue	Via Gateways
NAT Router	Yes
Peripheral Ports	USB, SD
LED Indicators	Power/Ready, Network
Reset Switch	Yes
Voice and Fax Codecs	G.711 A-law/U-law, G.722, G.723.1, G.726, G.729A/B, iLBC, GSM, AAL2-G.726-32
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP (RFC3261), STUN, SRTP, TLS
Media Encryption	SRTP, TLS
Universal Power Supply	Output: 12VDC, 2A; Input: 100 ~ 240VAC, 50 ~ 60Hz
Maximum Call Capacity	Supports up to 200 registered SIP devices/users Concurrent SIP calls: Up to 80 or 50 if SRTP encrypted