FIBERMEVoiPSPECALIST
Training

FVS

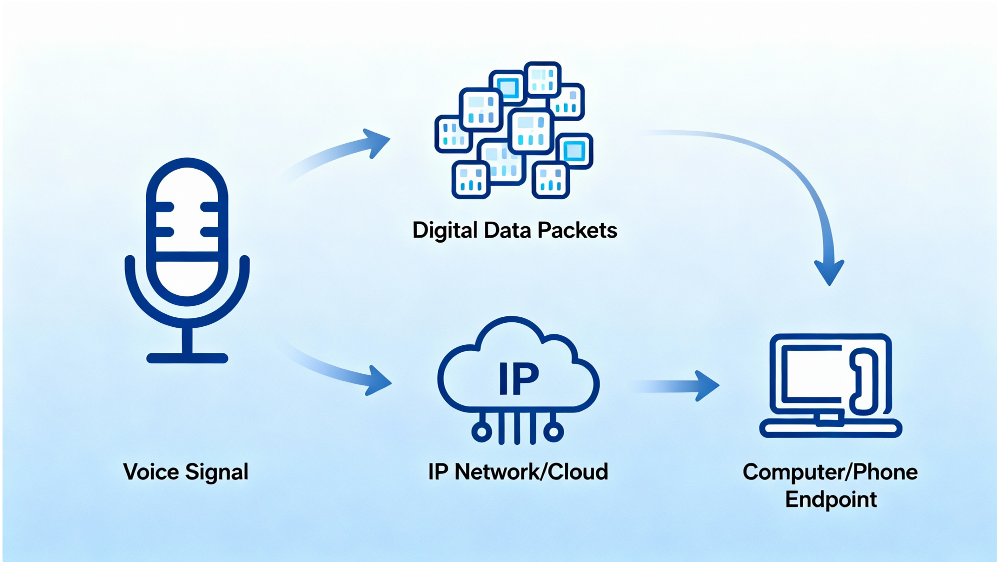
FIBERME
COMMUNICATIONS

Agenda

- What is VoIP?
- History and Evolution
- VoIP vs Traditional PSTN
- Benefits and Advantages
- Common Applications
- VoIP System Components
- VoIP Protocols
- Common Features
- Basic Call Center Features
- IP Phones
- Softphone
- Gateways
- Headsets
- FRC
- Use Cases

What is VoIP?

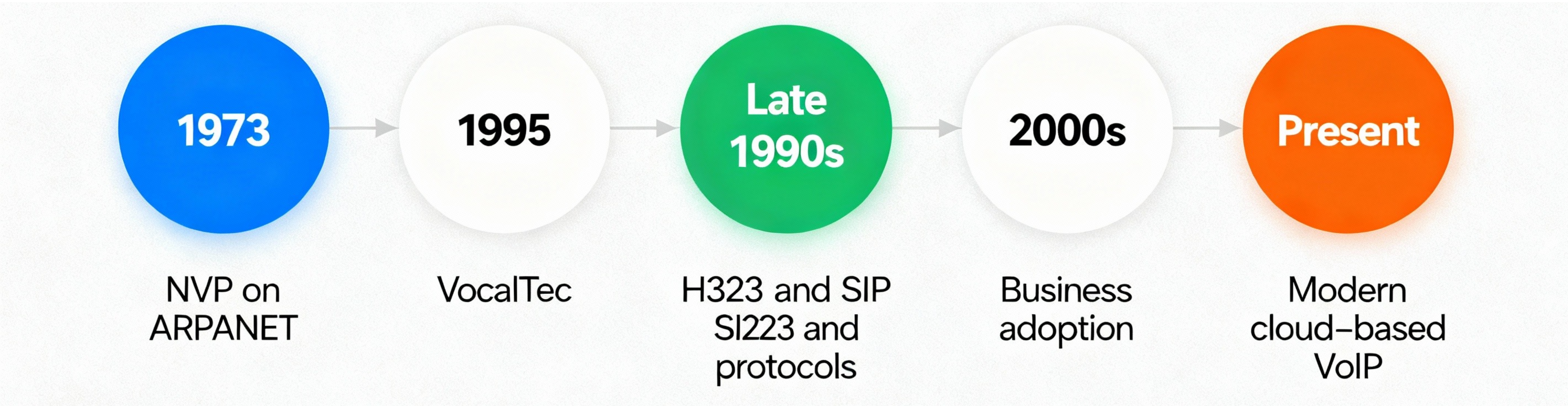
Voice over Internet Protocol (VoIP) is a technology that enables voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Instead of using traditional circuit-switched telephone networks (PSTN), VoIP converts voice signals into digital data packets and transmits them over IP networks



History and Evolution

VoIP technology has evolved significantly since its inception:

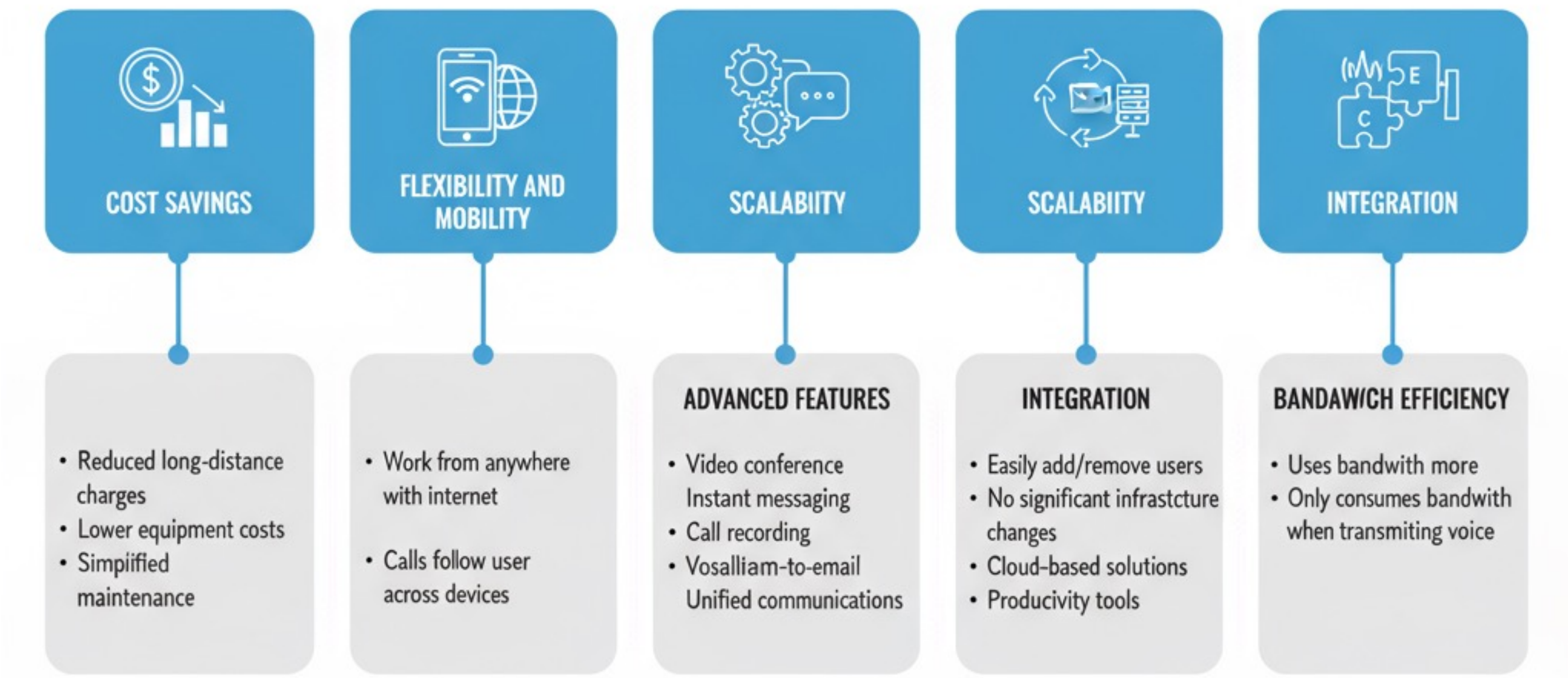
- 1973: Network Voice Protocol (NVP) developed for ARPANET
- 1995: First consumer VoIP software (VocalTec) introduced Late
- 1990s: Development of H.323 and SIP protocols
- 2000s: Widespread adoption by businesses and consumers. Present: Cloud-based VoIP, integration with unified communications, and mobile VoIP applications



VoIP vs Traditional PSTN

Feature	VoIP	Traditional PSTN
Technology	Packet-switched (digital)	Circuit-switched (analog/digital)
Cost	Lower operational costs	Higher infrastructure costs
Flexibility	Highly flexible, location-independent	Fixed location-based
Features	Advanced features (video, messaging, conferencing)	Basic voice calling
Scalability	Easy to scale	Difficult and expensive to scale
Infrastructure	Uses existing IP networks	Requires dedicated telephone lines

Benefits and Advantages



Common Applications

VoIP technology is deployed across various sectors

- Business Communications: Enterprise phone systems, call centers, and unified communications platforms.
- Residential Services: Home phone replacement services
- Mobile Applications: Smartphone softphone apps for remote workers
- Contact Centers: Advanced call routing, IVR systems, and analytics

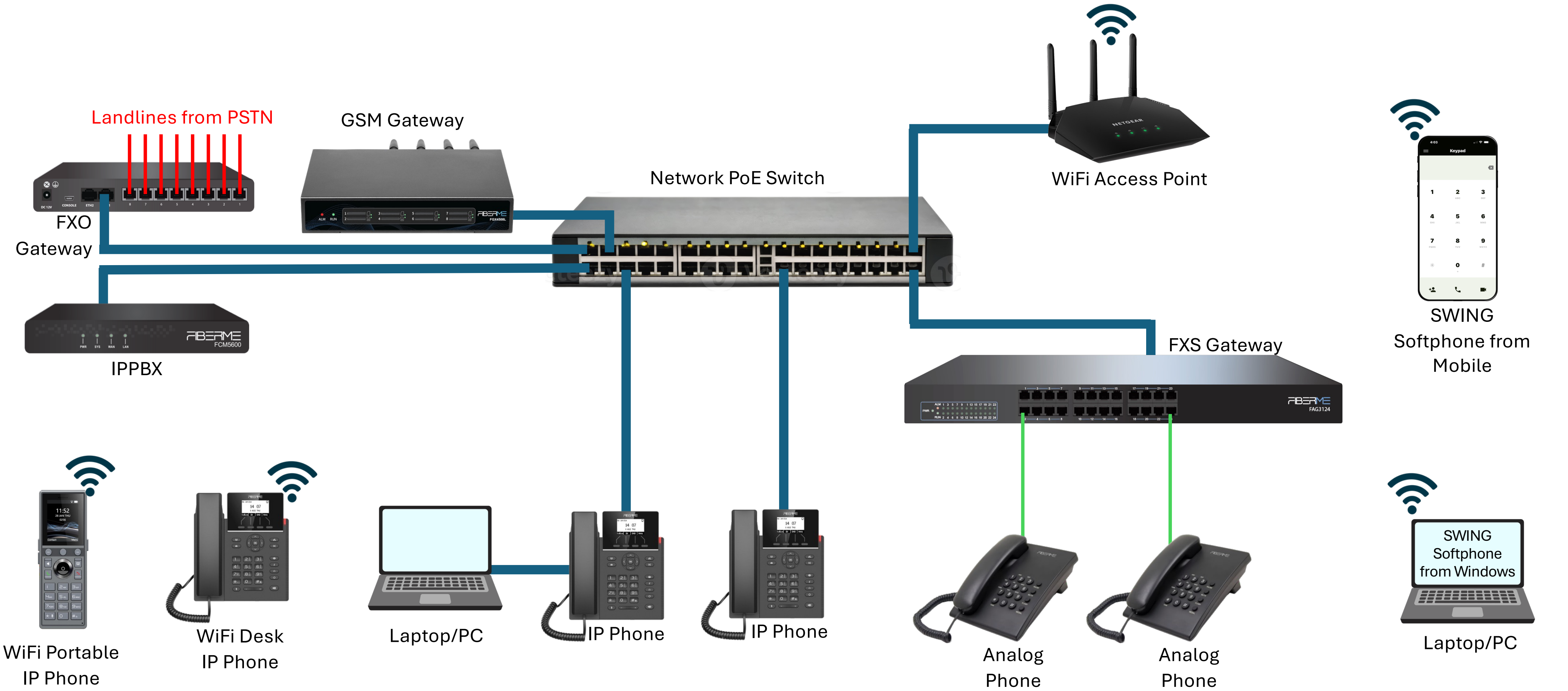
Business
Communications (Enterprise Systems)

Residential
Home Phone Services

Mobile Softphone Apps
(Remote Workers)

Contact Centers
(IVR & Call Routing)

VoIP System Components Topology



FIBERME COMMUNICATIONS

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VoIP System Components

Network/Internet

Network Switch

WiFi Access Point

IPPBX

IPPBX

Trunks

GSM Gateway

FXO Gateway

IPPBX

IP Phone

WiFi Desk IP Phone

WiFi Portable IP Phone

Analog Phone

Laptop/PC

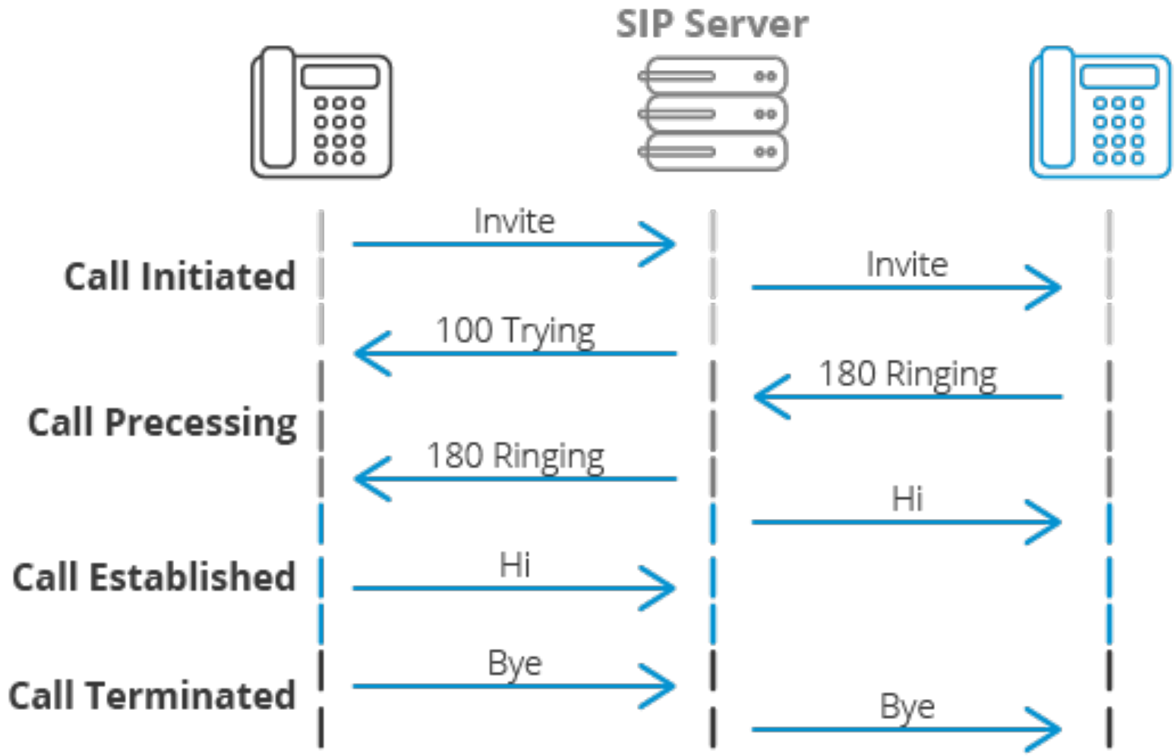
Softphone from Mobile

VoIP Protocols

SIP

Session Initiating Protocol

SIP is used to initiate, maintain, and terminate VoIP sessions. It handles call setup and teardown, managing who talks to whom and when the call should end. SIP uses port 5060 by default.



VoIP Protocols

RTP

Real-time Transport Protocol

RTP is used to transmit the actual voice data during a call. It handles the delivery of audio streams in real-time and ensures proper sequencing of packets. RTP is connection-oriented and optimized for low latency.

Audio Codec

A codec (Coder-Decoder) converts analog voice signals into digital data for transmission over IP networks, then back into sound at the receiver's end. Codecs determine call quality, required bandwidth, and latency.

Most use Codec

Bitrate	Bitrate
G.711	64 kbps
G.729	8 kbps
G.722	64 kbps
iLBC	13.3 / 15.2 kbps

G.711 is the most use Codec for the LAN

Call Latency

Delay in voice transmission. Should be less than 150ms for good quality.

Low Latency	Acceptable Latency	High Latency	Max Latency
20 to 50 ms	150 ms or Lower	150 ms to 250 ms	250 ms and Above

VoIP System Components | IPPBX

A complete VoIP system consists of several interconnected components that work together to enable voice communication over IP networks

IPPBX, is the central control elements in VoIP systems . Unlike traditional hardware-based PBX systems, softswitches are software applications running on standard servers.

Here is FIBERME FCM IPPBX Series, FCM is a definition for (FIBERME Call Manager)

	FCM630A	FCM5600	FCM5404	FCM7030	FCM7250
SIP Users	250	500	60-300	300-1500	2500
Concurrent Calls	50	75	15-60	50-175	275
FXO Ports	-	-	4	-	-
FXS Ports	-	-	2	-	-
Network Ports	Dual Network Ports	Dual Network Ports	Dual Network Ports	Dual Network Ports	Dual Network Ports
Communication Platform	Asterisk	Asterisk	FreeSwitch	Asterisk	Asterisk

VoIP System Components | IPPBX

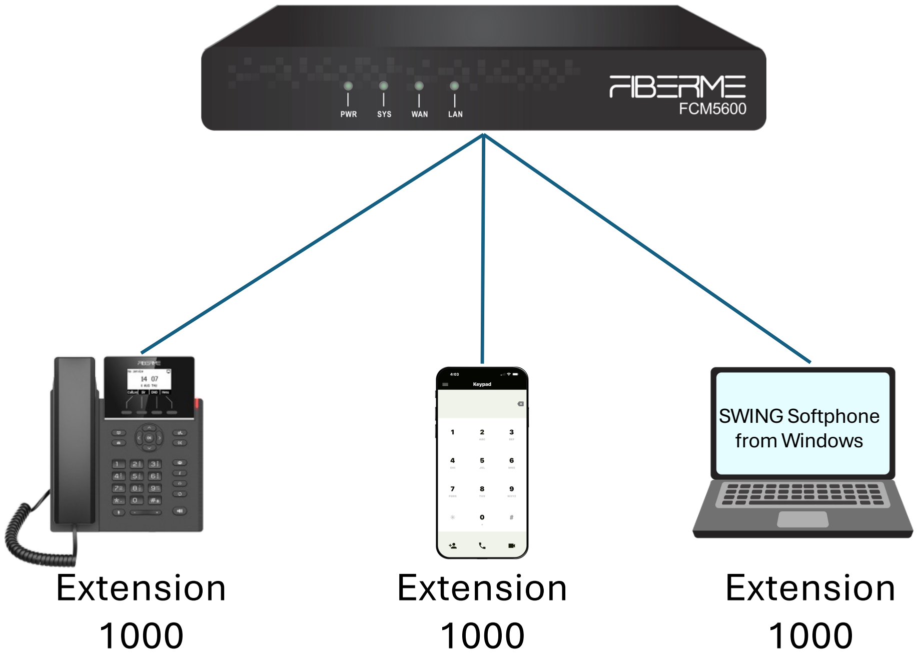
FCM IPPBX Series Common Features

- Voicemail
- Call Forward
- Caller ID
- Ring Group
- IVR
- Call Queue
- Call Reporting
- Call Recording
- SIP Trunks
- DISA
- Inbound Routes
- Outbound Routes
- Office Time
- Paging/Intercom
- Voice Prompt
- Feature Codes
- Parking Lot
- Callback

VoIP System Components | IPPBX | Main Features

Concurrent Registration

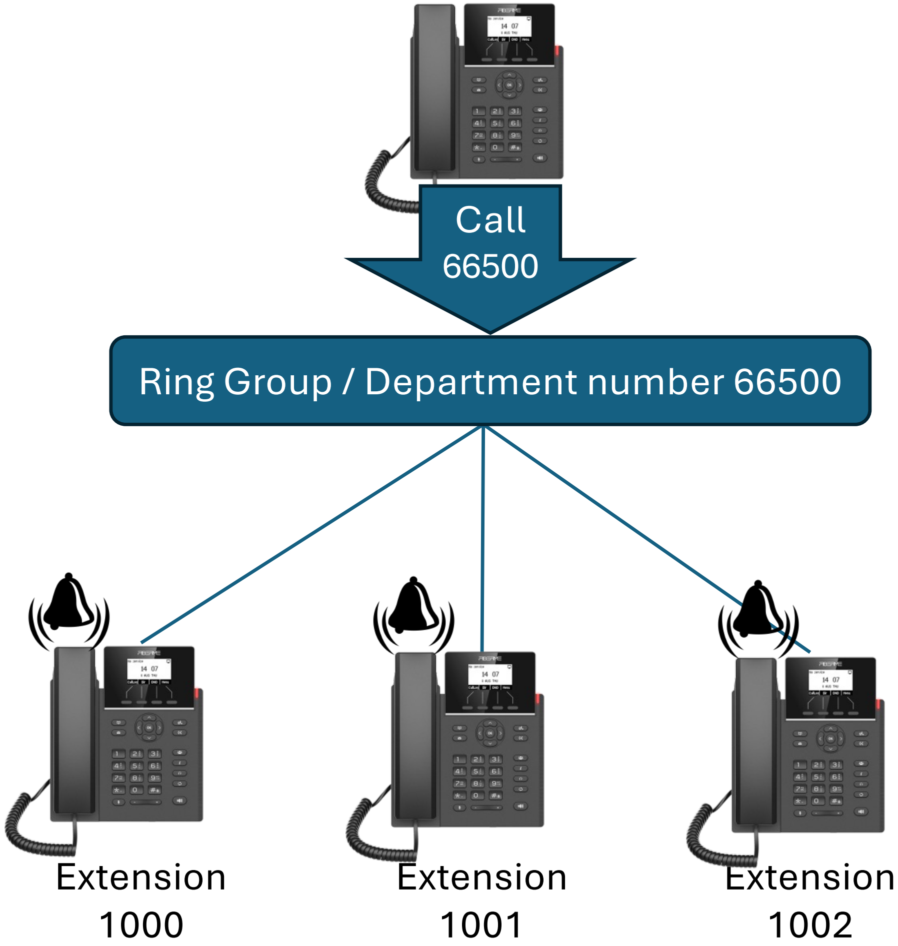
- FIBERME FCM IPPBX Series supports multi-endpoint registration for extensions. You can register the same extension on more than one device simultaneously (up to 10 devices on some models):
- Register from multiple IP Phones or a combination of IP Phones and softphones.
- All registered endpoints can make, receive, and manage calls for the same extension.
- This provides flexible connectivity for users needing access on several devices, whether at their desk, on their mobile.



VoIP System Components | IPPBX | Main Features

Ring Group / Department

- A Ring Group or Department serves the same function in the PBX system: it allows users to call a specific number assigned to a group (such as Sales, Support, or a department), and that call will ring multiple extensions at the same time.
- You can assign several extensions to a ring group or department.
- When someone calls the group number, all assigned extensions ring simultaneously, and any member can answer the call.
- This setup is ideal for teams or departments where availability must be maximized and any available member can respond to incoming calls.



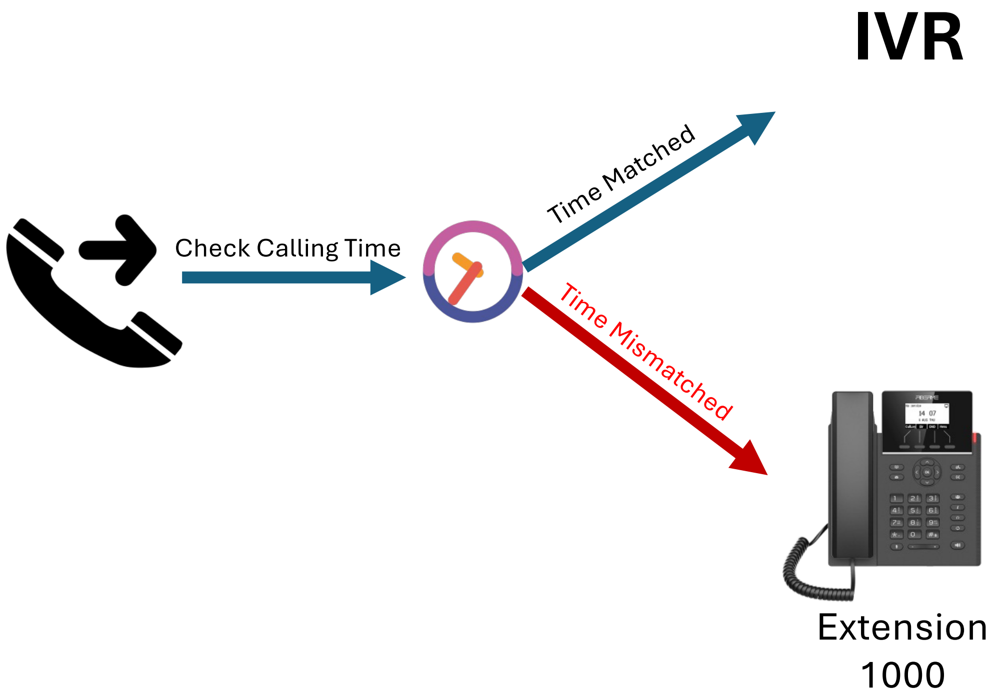
VoIP System Components | IPPBX | Main Features

Time Condition Routing

Time Condition Routing is a PBX feature that organizes incoming calls by routing them according to the current time or schedule.

- You set time conditions (work hours, after hours, holidays, etc.).
- If a call matches the defined time (e.g., during business hours), it will be routed to the IVR menu or any specified destination.
- If the call does not match (e.g., after hours), it can be routed to another destination, such as extension 1000, voicemail, or a recording.

This ensures calls are handled differently based on time, improving both customer experience and business efficiency.



VoIP System Components | IPPBX | Main Features

DISA

DISA (Direct Inward System Access) is a PBX feature that allows external callers to access internal PBX features such as making outbound calls, transferring calls, or accessing specific functions as if they were calling from within the organization.

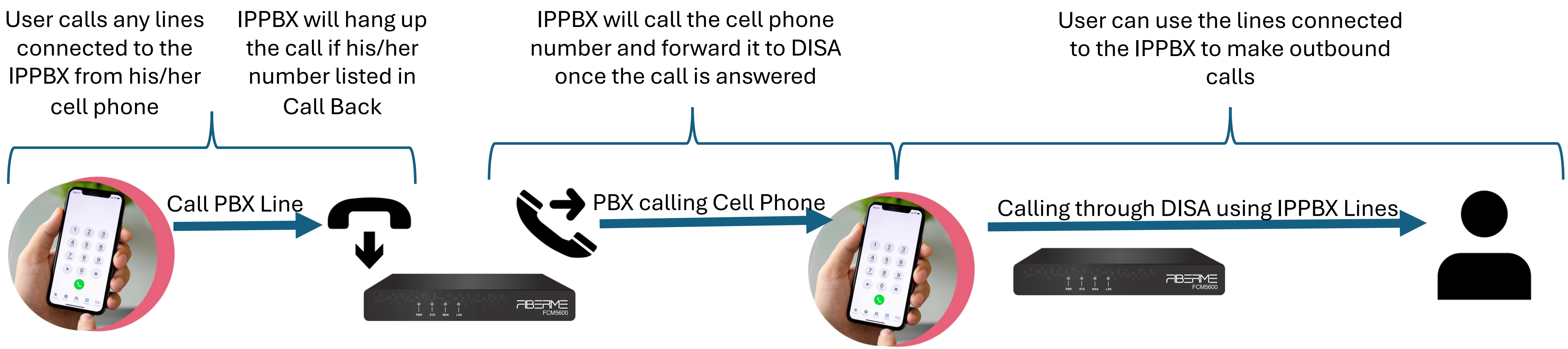
- The caller dials a special DISA number or menu option.
- After entering a security PIN, the caller gets dial tone and can place calls or invoke system features.
- Commonly used to enable remote workers or executives to make outbound calls using the company's phone lines, ensuring calls appear from the company number rather than their personal phone.

DISA provides convenience and flexibility but must be secured with strong authentication (like PIN codes) to prevent abuse or unauthorized call routing through your system.

VoIP System Components | IPPBX | Main Features

Call Back

The callback feature is primarily intended for users who frequently make long-distance or international calls from their mobile phones, which can result in high service charges.
By using callback, calls are routed and connected via trunks on the IP-PBX rather than directly through the mobile network or a landline, allowing users to avoid premium charges for long distance or international calls and significantly reduce overall call costs.

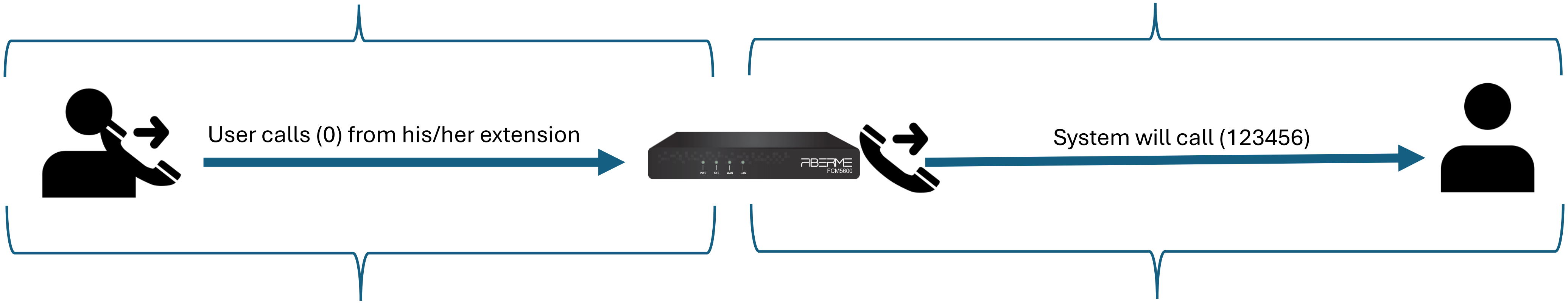


VoIP System Components | IPPBX | Main Features

Speed Dial

Speed Dial is a telephony feature that allows users to quickly call predefined numbers by pressing a short sequence or single button, instead of dialing the full phone number.

User can set Speed Dial for the number (0) to call a specific number (123456)

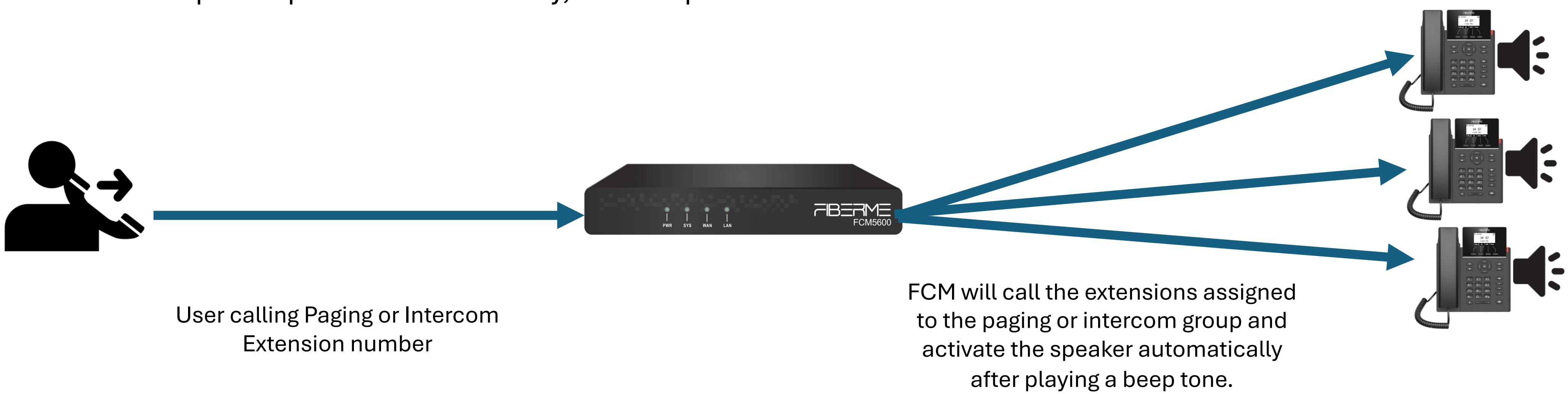


VoIP System Components | IPPBX | Main Features

Paging & Intercom

Paging & Intercom are telephony features provided by PBX systems, for organizational announcements and direct communication.

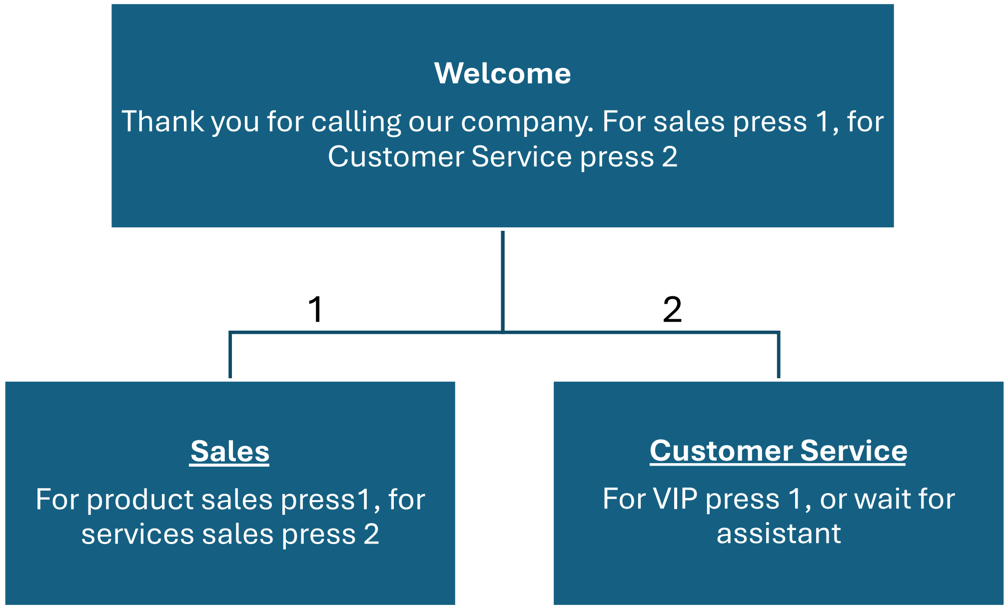
- **Paging:** Lets a user broadcast a one-way audio message to multiple extensions or devices simultaneously (e.g., “all hands meeting now”). Recipients hear the announcement but cannot reply through the paging session.
- **Intercom:** Initiates a two-way audio connection between two extensions, turning the recipient’s device into a hands-free speakerphone automatically, so both parties can talk back and forth.



VoIP System Components | IPPBX | Main Features

Call Center | IVR

IVR menus are automated phone systems that guide callers through options using keypad inputs. Callers can select from predefined choices like "Press 1 for sales" to get information or be routed to the right department without talking to a person. This helps improve call efficiency, allows self-service, and reduces wait times. IVR menus use prerecorded or generated voice prompts and can include multiple layers of options for different services.



VoIP System Components | IPPBX | Main Features

Call Center | Queue

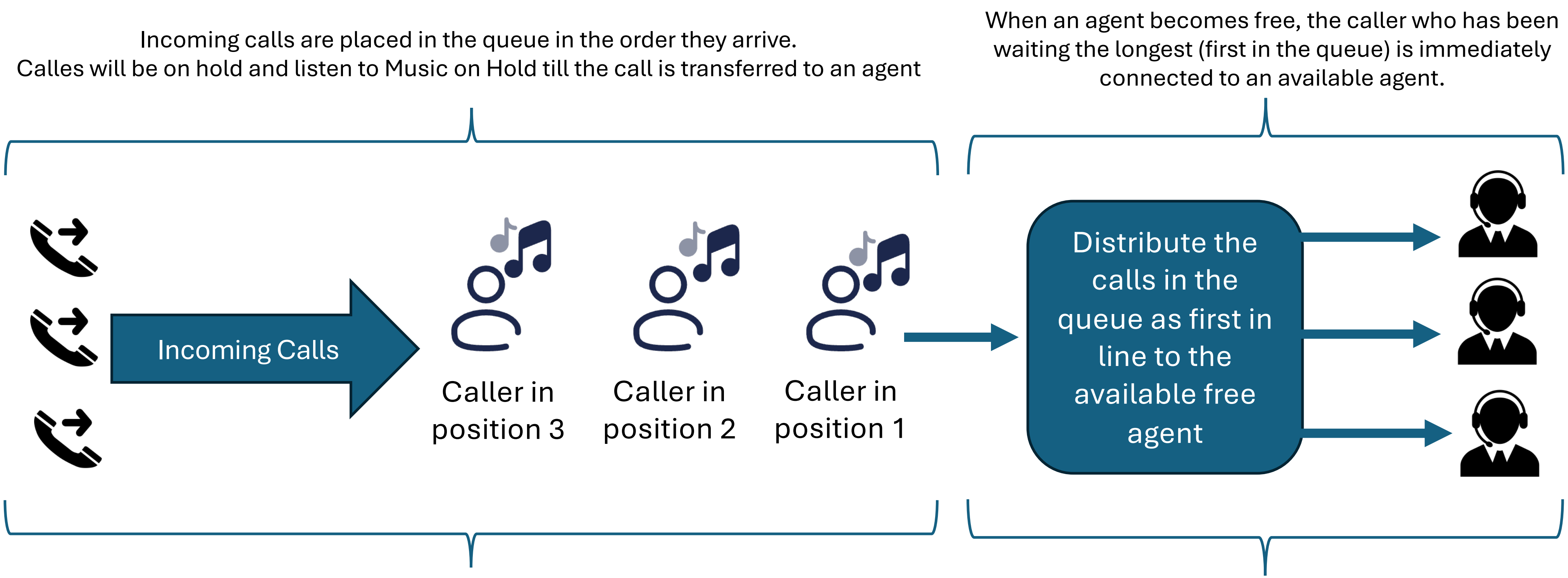
The main purpose of the Queue is to manage incoming calls directed to the same department based on the FIFO (First In, First Out) principle.

- When multiple callers dial in and all agents are busy, each caller is placed in a queue according to their arrival time.
- The first caller to enter the queue (the one who has been waiting the longest) will be the first to be connected to the next available agent.
- This process continues for each subsequent caller in the order of their arrival.

FIFO guarantees fairness, ensuring callers are helped based on how long they've waited, not on any other priority or criteria. It’s the standard queuing method in call centers for managing and distributing incoming calls.

VoIP System Components | IPPBX | Main Features

Call Center | Queue



VoIP System Components | IPPBX | Main Features

Call Center | Queue, Call Distribution Strategy

RING ALL

All available agents' phones ring at the same time for each new call. The first to answer is connected, and the others stop ringing.

ROUND ROBIN

Calls are routed in a rotating order among agents, distributing calls as evenly as possible.

LEAST RECENT

Assigns the next call to the agent who has been idle the longest.

RANDOM

Calls are assigned to agents at random from the pool of free agents.

FEWEST CALLS

The agent who has answered the least number of calls recently gets the next call, helping balance workloads among all agents.

LINEAR

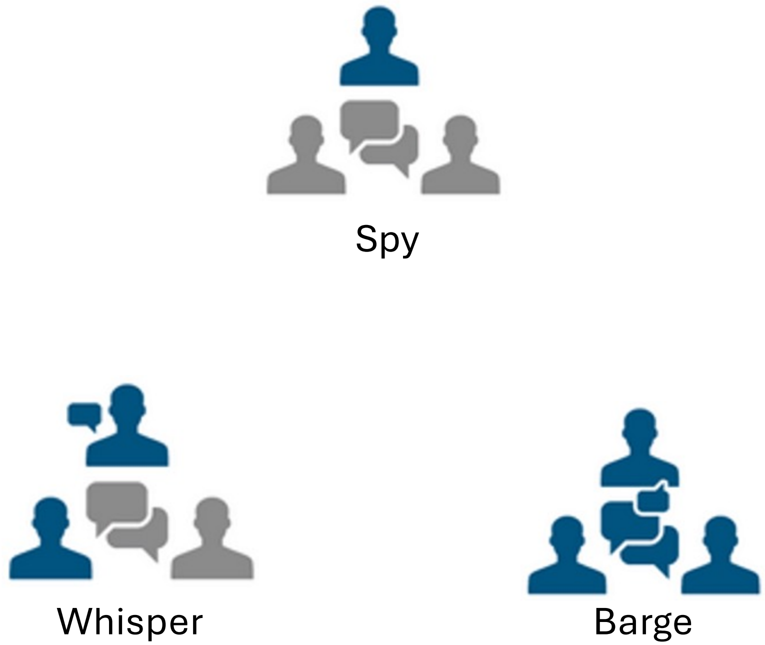
Calls are always offered to agents in a fixed, pre-set order—starting from the first agent each time, moving through the list one by one.

VoIP System Components | IPPBX | Main Features

Call Center | Call Barge

Call Barge is a PBX feature that allows a supervisor or authorized user to actively join an ongoing call between an agent/extension and another party.

- **Spy:** Lets supervisors or authorized users listen in on active calls between agents/extensions and other parties without participating or being heard.
- **Whisper:** Allows a supervisor to speak directly to an agent during an active call without the external caller hearing the conversation.
- **Barge:** Lets a supervisor or authorized user join an active call between an agent and another party instantly turning the call into a three-way conversation.



VoIP System Components | IPPBX | Main Features

Call Center | CDR (Call Details Record)

It is a comprehensive report that provides detailed information about all call activities.

Key components of CRD:

- Call Status
- Caller Number
- Callee Number
- Type (Inbound/Outbound)
- Used Trunk
- Call Duration

STATUS	CALL FROM	CALL TO	ACTION TYPE	START	END	DURATION	RECORDED FILE
ANSWERED	30	[redacted]770	Outbound[4301]	12-11-2025 12:14:57 PM	12:15:12	00:00:08	No recorded file
ANSWERED	[redacted]770	99700	Inbound[4301]	12-11-2025 12:14:22 PM	12:14:22	00:00:00	No recorded file
ANSWERED	30	[redacted]067	Outbound[4301]	12-11-2025 12:12:04 PM	12:12:35	00:00:28	No recorded file
ANSWERED	[redacted]50	99700	Inbound[4301]	12-11-2025 12:11:18 PM	12:11:18	00:00:00	No recorded file
NO ANSWER	[redacted]50	30	Inbound[4301]	12-11-2025 12:10:57 PM	12:11:02	00:00:00	No recorded file
NO ANSWER	30	[redacted]050	Outbound[4301]	12-11-2025 12:09:22 PM	12:09:23	00:00:00	No recorded file
ANSWERED	30	[redacted]050	Outbound[4301]	12-11-2025 12:09:04 PM	12:09:17	00:00:11	No recorded file
NO ANSWER	30	[redacted]50	Outbound[4301]	12-11-2025 12:08:40 PM	12:08:40	00:00:00	No recorded file
ANSWERED	[redacted]770	30	Inbound[4301]	12-11-2025 12:06:58 PM	12:07:12	00:00:12	No recorded file
ANSWERED	[redacted]770	30	Inbound[4301]	12-11-2025 12:06:26 PM	12:06:52	00:00:21	No recorded file

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VoIP System Components | IPPBX | Main Features

Call Center | Call Recording

FCM enables recording of all call types, including outbound, inbound, and internal calls between users. Call recording can be configured based on:

- Specific extension
- Specific trunk or line
- Specific queue

The Standard recording format is WAV, it costs ~0.85 MB per minute or each call.

Example:

5-minute call = ~0.85 MB per minute x 5 minutes = 4.25 MB

1GB = ~20 Hours of talking time recording

VoIP System Components | Endpoints

End Points

A SIP Extension can be registered and used on various endpoint types, including:

- IP Phones: Hardware desk phones with network connectivity.
- Analog Phones: Traditional telephones connected via an analog adapter or FXS gateway.
- Softphones: Software-based phones available for Windows, Mac, mobile devices (iOS/Android), and web browsers.

IP Phone

Analog Phone

Softphone

IP Phones

VoIP System Components | Endpoints

Entry Level IP Phones

FAP2714

	FAP2714P	FAP2714G	FAP2714W
SIP Accounts	2		
Screen	2.4” Graphical Screen with Backlit		
Ethernet Ports	Dual Network Ports		
PoE	Integrated PoE		-
Bandwithe	100Mbps	1Gbps	100Mbps
HD Voice Qulity	Yes		
WiFi	-		Dual Band WiFi
PSU Included	No		Yes

VoIP System Components | Endpoints

Mid-level IP Phones

FAP2733

<table>
<tr><th></th><th>FAP2733P</th><th>FAP2733G</th><th>FAP2733W</th></tr>
<tr><td>SIP Accounts</td><td colspan="3">4</td></tr>
<tr><td>Line Kees</td><td colspan="3">3</td></tr>
<tr><td>Screen</td><td colspan="3">2.5” Color Screen</td></tr>
<tr><td>Ethernet Ports</td><td colspan="3">Dual Network Ports</td></tr>
<tr><td>PoE</td><td colspan="2">Integrated PoE</td><td>-</td></tr>
<tr><td>Bandwithe</td><td>100Mbps</td><td>1Gbps</td><td>100Mbps</td></tr>
<tr><td>HD Voice Qulity</td><td colspan="3">Yes</td></tr>
<tr><td>WiFi</td><td colspan="2">-</td><td>Dual Band WiFi</td></tr>
<tr><td>PSU Included</td><td colspan="2">No</td><td>Yes</td></tr>
</table>

VoIP System Components | Endpoints

Enterprise IP Phones

FAP2760

	FAP2760
SIP Accounts	20
Line Keys	10
Screen	4.3” Color Screen
Ethernet Ports	Dual Network Ports
PoE	Integrated PoE
DSS Keys	45 DSS
Bandwithe	1Gbps
HD Voice Qulity	Ultra HD Voice Quality
Support Expahnsion	Yes (FX60), Up to 5 Expehnsions
PSU Included	No

VoIP System Components | Endpoints

	FW620
SIP Accounts	4
Screen	2” Color Screen
Network	Dual Band WiFi
WiFi Roaming	Supported
Push to Talk	Yes
DSS Keys	16 Soft DSS
Dropfree	1.8m Dropfree
HD Voice Qulity	Ultra HD Voice Quality
Base Charger	Yes
Charger Type	USB Type-C
Loud Speaker	Yes

Portable WiFi IP Phones

FW620

VoIP System Components | Endpoints

Analog Phone

The FIBERME A200 is a robust and dependable analog phone, meticulously designed for seamless compatibility with any analog PBX system. This feature makes it an exceptional choice for a diverse array of office environments, from small businesses to large corporate settings. The phone comes equipped with essential features like redial, mute, and volume control, ensuring that it meets the basic telecommunication needs of any workplace with ease and efficiency.

A200

VoIP System Components | Endpoints

Softphones

swing

SWING for Windows

SWING for Mobile

VoIP System Components | Endpoints

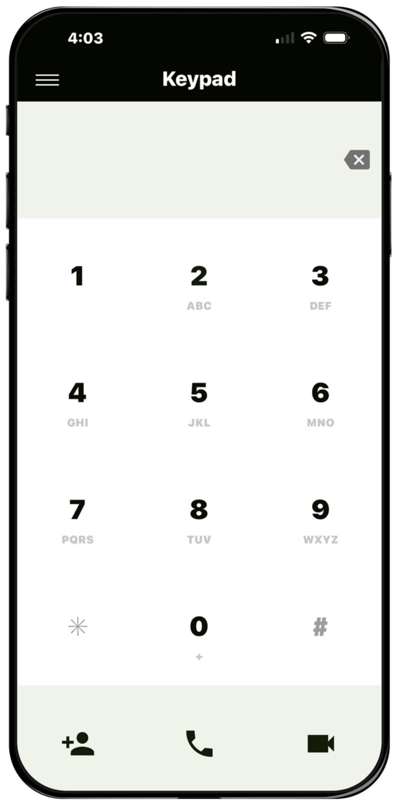
swing

SWING Lite for IOS

Support UDP	√
Support TCP	√
Support TLS	√
Loud Speaker	√
Support Apple Call kit	√

Push Notification	√
SRTP	√
STUN	√
ICE	√
NAT	√

Call Control	Mute, Hold/Resume, DTMF, Transfer (Attend/Blind)
Conference Call	3 Ways Conference Call
Codec	GSM, PCMA, PCMU, iLBC, Speex, G729, G722, G721, G723, OPUS,

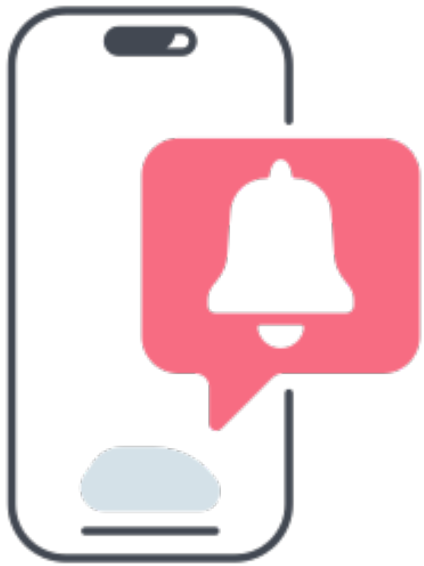


VoIP System Components | Endpoints

swing

FIBERME SWING Lite softphone for IOS

Push Notification



Push notifications enable the softphone to operate in the background on mobile devices, even when the app is closed.

VoIP System Components | Endpoints

swing

SWING for Windows

Support UDP	√
Support TCP	√
Support TLS	√
Loud Speaker	√
Support Apple Call kit	√

Push Notification	√
SRTP	√
STUN	√
ICE	√
NAT	√

Call Control	Mute, Hold/Resume, DTMF, Transfer (Attend/Blind)
Conference Call	3 Ways Conference Call
Codec	GSM, PCMA, PCMU, iLBC, Speex, G729, G722, G721, G723, OPUS,

VoIP System Components | Gateways

Gateways

FIBERME offers three types of Gateways: FXO, FXS, and GSM. These gateways enable your communication call manager (IP PBX) to connect seamlessly with a wide range of devices and trunk lines:

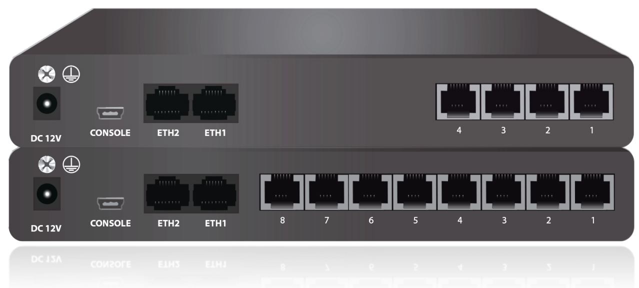
- **FXO Gateway:** Connects your PBX system to analog PSTN trunk lines, allowing calls to and from traditional telephone networks.
- **FXS Gateway:** Connects analog phones or fax machines to your IP PBX, letting you use legacy devices with modern VoIP systems.
- **GSM Gateway:** Connects your PBX system to SIM Cards, allowing calls to and from GSM operators using Cell Phone Numbers.

These gateways expand the integration options of your IP PBX, supporting flexible connectivity for various telecommunications environments.

VoIP System Components | FXO Gateways

FXO Gateway: Connects your PBX system to analog PSTN trunk lines, allowing calls to and from traditional telephone networks. FIBERME provides 2 models of FXO Gateways:

FAG4104:	4 FXO Ports
FAG4108:	8 FXO Ports

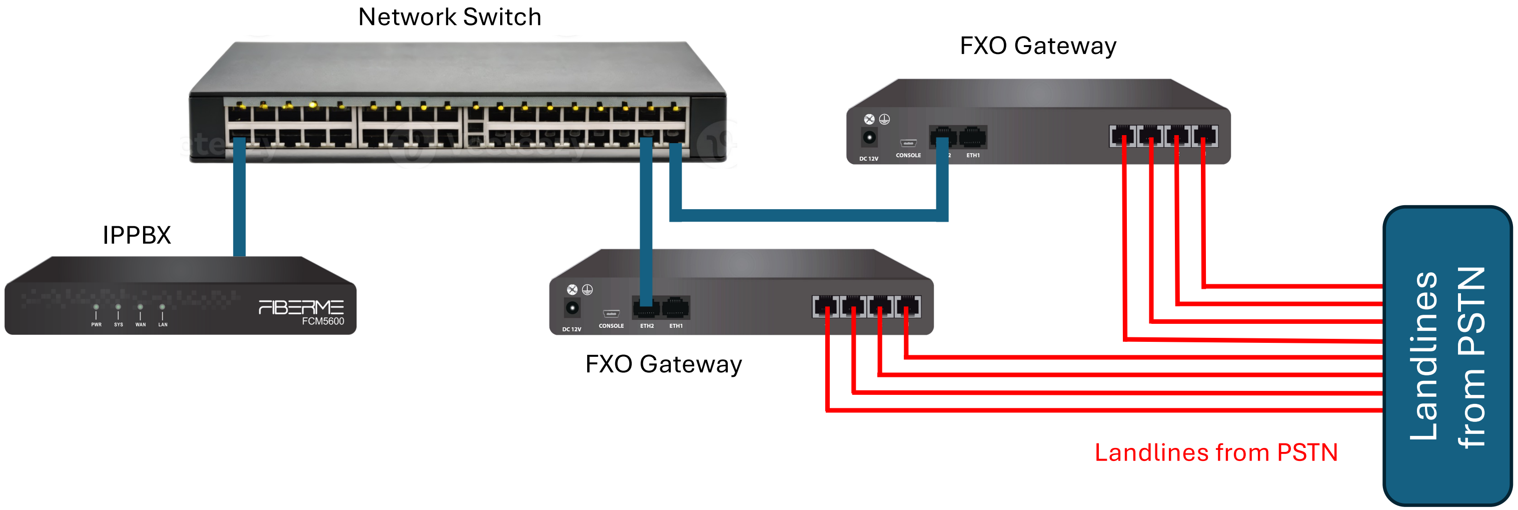


FIBERME FXO Gateway is SIP Standard and 100% compatible with any SIP Server and most use IPPBX and more:

VoIP System Components | FXO Gateways

User can connect more than FXO or/and GSM Gateway in the same time

FXO Gateway connected physically to the Network Switch, not to the IPPBX directly



VoIP System Components | GSM Gateways

GSM Gateway: Connects your PBX system to SIM Cards, allowing calls to and from GSM operators using Cell Phone Numbers.

FGX4508G:	8 SIM, 2G
FGX4508L:	8 SIM, 4G

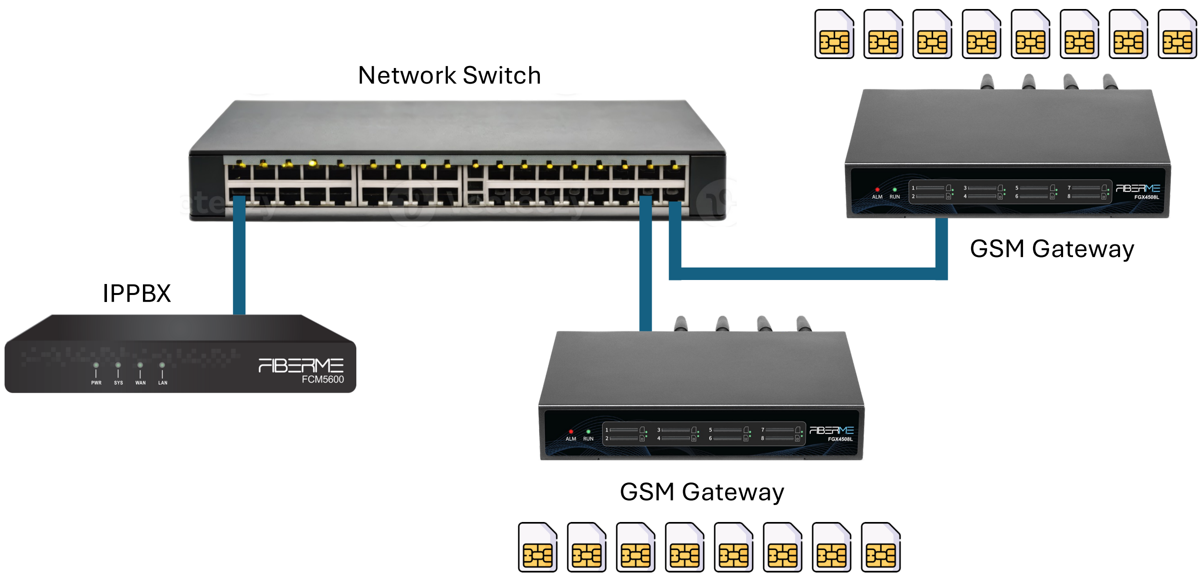
FIBERME GSM Gateway is SIP Standard and 100% compatible with any SIP Server and most use IPPBX and more:

CISCO elastix DINSTAR GRANDSTREAM 3CX Yeastar XonTel Issabel XORCOM

VoIP System Components | GSM Gateways

User can connect more than GSM or/and FXO Gateway in the same time

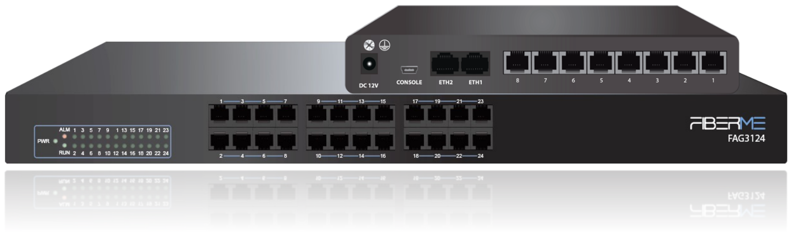
GSM Gateway connected physically to the Network Switch, not to the IPPBX directly



VoIP System Components | FXS Gateways

FXS Gateway: Connects analog phones or fax machines to your IP PBX, letting you use legacy devices with modern VoIP systems. Only analog phones can be connected to the FXS Gateway, while digital phones are not supported

FAG3108:	8 FXS Ports
FAG3124:	24 FXS Ports



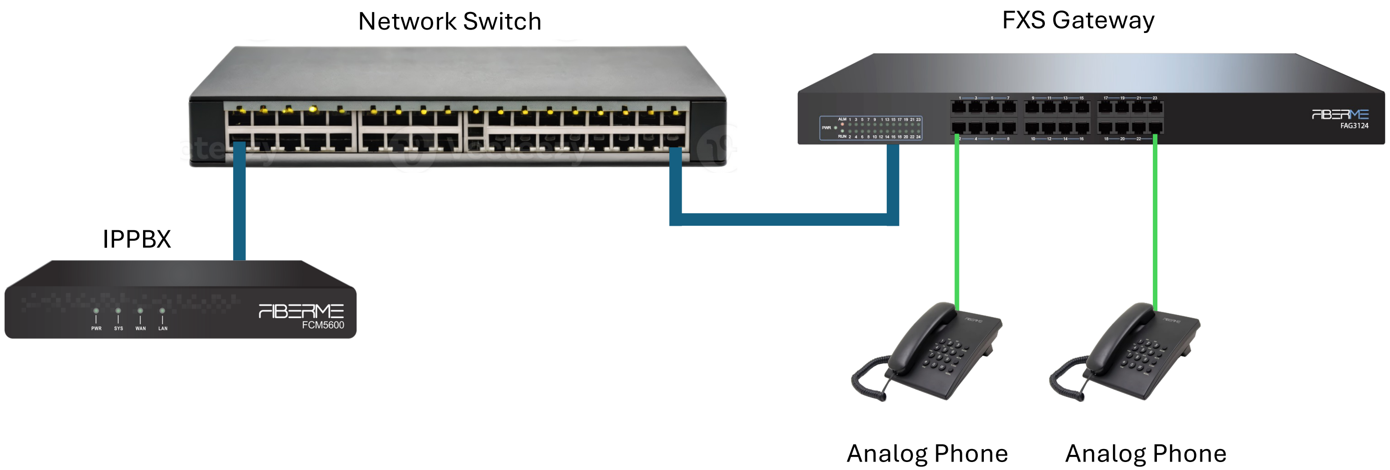
FIBERME FXS Gateway is SIP Standard and 100% compatible with any SIP Server and most use IPPBX and more:

CISCO elastix DINSTAR GRANDSTREAM 3CX Yeastar XonTel Issabel XORCOM

VoIP System Components | FXS Gateways

User can connect more than FXS Gateway in the same time

FXS Gateway connected physically to the Network Switch, not to the IPPBX directly, then connect analog phones to the analog ports located in FXS Gateway.



FIBERME COMMUNICATIONS

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HEADSETS

VoIP System Components | Headsets

FCH7201

Wired Headset

	FCH7201	FCH7201D
Design	Mono	Duo
Noise Reduction Method	Software Noise Cancellation	
Microphone Array	1 Mic Array	
Speaker Size	28.0 mm	
Speaker Sensitivity	103±3dB	
In-line Controls	Answer/End a Call Button, Volume Up/Down Button, Microphone Mute/Unmute Button	
Interface	USB Type-A	

VoIP System Components | Headsets

FCH7110

Wired Headset

	FCH7110	FCH7110D
Design	Mono	Duo
Noise Reduction Method	AI mic noise-cancelling technology	
Microphone Array	3 Mics Array	
Speaker Size	27.0 mm	
Speaker Sensitivity	101±1dB SPL @ 1 kHz	
In-line Controls	Answer/End a Call Button, Volume Up/Down Button, Microphone Mute/Unmute Button	
Interface	USB Type-A	

VoIP System Components | Headsets

FBH35D Bluetooth Headset

Design	Duo
Noise Reduction Method	AI mic noise-cancelling technology
Microphone Array	2 Mics Array
Bluetooth	V5.2
Charging Time	2 Hours
Standby Time	200 Hours
Talking Time	45 Hours (70% Volume)
Charching Base	Yes
Charging Type	USB Type-C
Battary Capacity	3.7V, 500mAh
Multi Funcation Keys	Play/Pause, Answer/Hang up, Volume(+/-), Power-on/Power-off

FRC (FIBERME Remote Connect) is a service offered by FIBERME that enables users, extensions, Gateways, and IPPBXs to connect and register with the IP PBX system from outside the local network.

Supported IPPBX
FCM5600 | FCM7030 | FCM7250

Supported Gateways
All FIBERME Gateways

Supported Softphone
FIBERM SWING, and SIP softphone

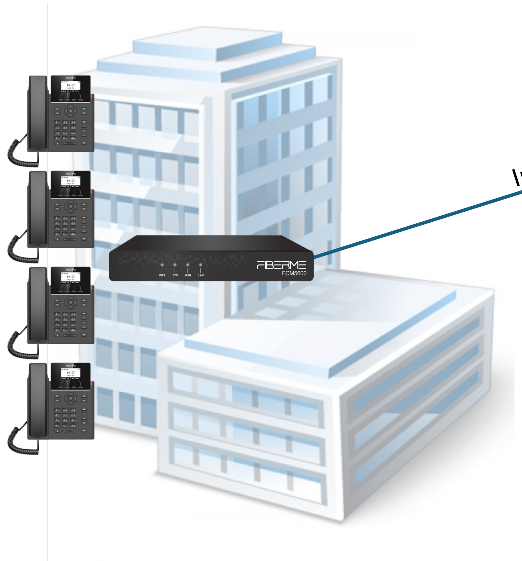
FRC

FIBERME Remote Connect

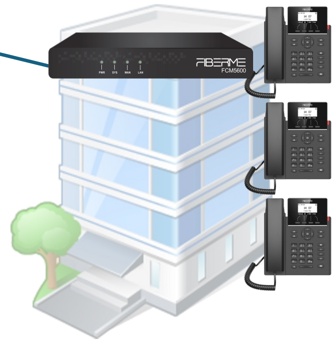
No Static IP, VPN, or even Dyn DNS Required

Seamless Communications

Seamless Communications



Business Office



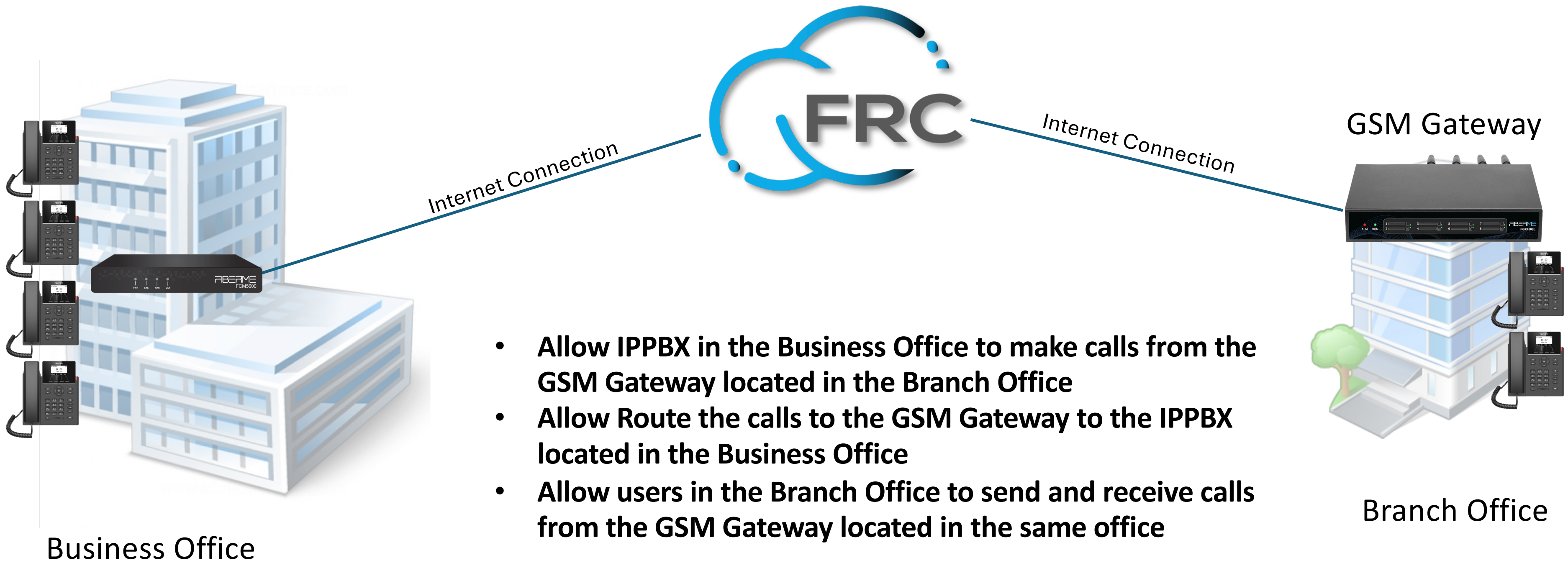
Branch Office

- **Allow making internal calls between offices**
- **Allow transferring calls to another office**
- **Allow Office to use lines and trunks from another office**
- **Allow conference calls between offices**

Seamless Communications

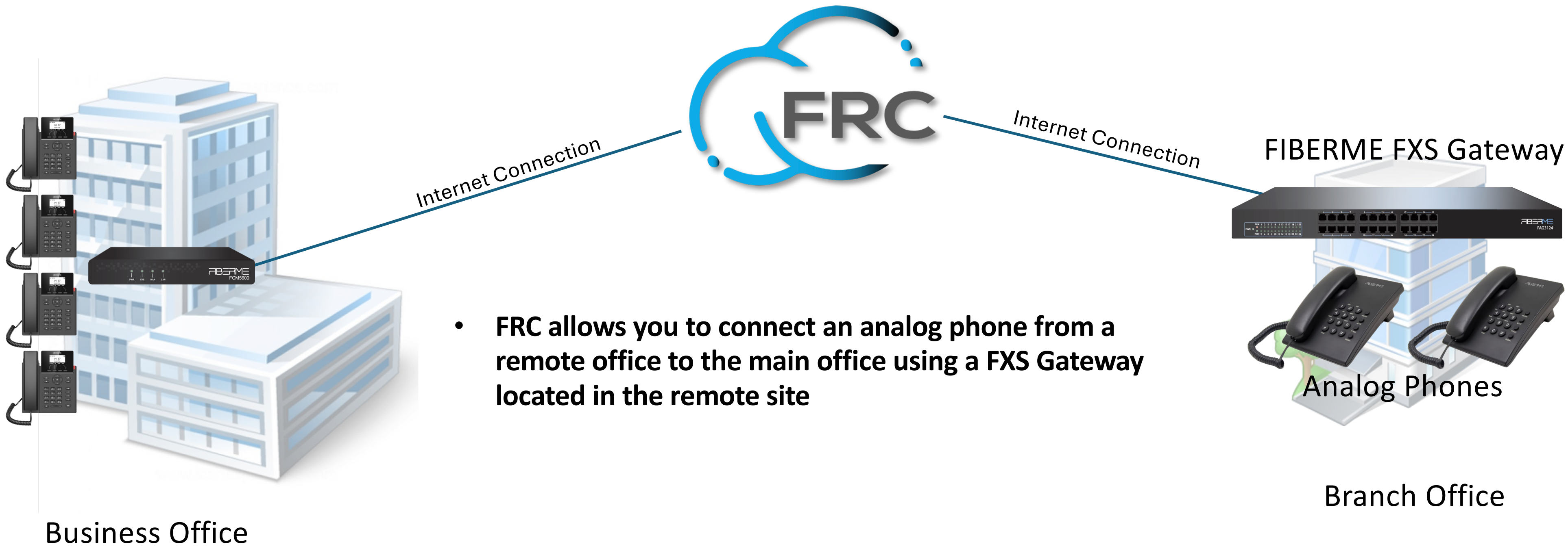
- **Allow Shops to call each other**
- **Allow Shops to call the Business Office**
- **Allow the Business Office to call the Shops**
- **Allow Shops to use Business Office trunk lines**
- **Allow Business office to transfer calls to Shops**
- **Allow making conference between shops and business office**

Seamless Communications



- **Allow IPPBX in the Business Office to make calls from the GSM Gateway located in the Branch Office**
- **Allow Route the calls to the GSM Gateway to the IPPBX located in the Business Office**
- **Allow users in the Branch Office to send and receive calls from the GSM Gateway located in the same office**

Seamless Communications

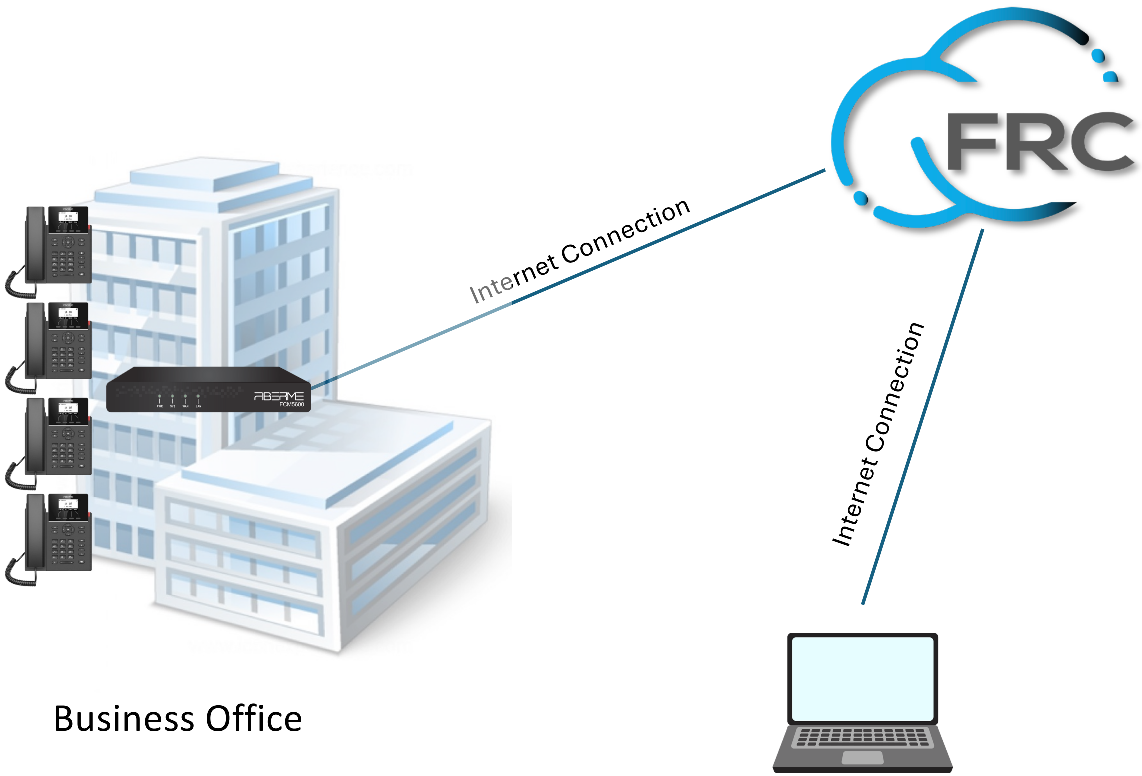


- **FRC allows you to connect an analog phone from a remote office to the main office using a FXS Gateway located in the remote site**

FIBERMEVoiPSPECALIST

Work from any remote site or event home using:

Work from any remote site or event home using:



REMOTE MANAGMENT

FIBERME FRC, allows users to access and manage PBX from anywhere.
The user can access the PBX GUI and get full control.

USE CASES

Use Cases

Is the internet is required to make internal calls between extensions located in the same local area network?

No

The Internet is not part of, and not required for, internal calls between extensions in the same local area network

FCM5600

Gransdtream IPPBX

Use Cases

Can the user interconnect the FIBERME FCM5600 IPPBX installed at the business office with the Grandstream IPPBX located at a remote site using FRC?

Yes

You can do that using FRC, and it'll allow the following features:

- Allow users in both IPPBXs to call each other internally.
- Allow users in Grandstream IPPBX to use Trunk on FIBERME FCM5600
- Allow users in FIBERME FCM5600 to use Trunk on Grandstream IPPBX
- Allow users to transfer calls to other IPPBX users.
- Allow users to make conference calls between users in both IPPBX.

FCM5600

FIBERME FGX4508 GSM Gateway

Use Cases

Can we connect FIBERME FCM5600 IPPBX located in Business Office IPPBX with FIBERME FGX4508 GSM Gateway located in another site using FRC?

Yes

Yes, all FIBERME Gateways can register to FCM5600 remotely using FRC

Use Cases

Can the Grandstream GRP IP Phone series be registered remotely to the FCM5600 using FRC?

Yes

All Grandstream GRP & GXV Series IP Phones can register to FCM5600 remotely using FRC

Use Cases

Can we connect 4 devices of FCM5600 IPPBXs located in different sites to each other remotely using FRC?

Yes

You can do that using FRC, as we can connect up to 50 devices of FCM5600 devices together remotely using FRC

Use Cases

Is FIBERME SWING Lite softphone for IOS required Internet to support Push Notification, even for extensions registered from the same Local Area Network

Yes

Push notifications require an active internet connection to reach the softphone. The notifications are delivered from FCM to the device through Apple servers, so the mobile must be connected to the internet to receive them.

FVS

FIBERME VoiP SPECALIST
Training

Thank you

FIBERME
COMMUNICATIONS