



**EMPATİQ®**  
iletişim teknolojileri

## Company Overview

- Founded in 2002
- Over 500 employees
- Product Portfolio contains over 40 products: *20 different IP Phones, 5 ATAs, 10 IP Surveillance cameras, 5 Video Encoder/Decoders & 2 IP PBXs*
- Primarily serving small-to-medium size businesses (SMBs) and consumer markets

### US



Boston - Headquarters  
Los Angeles, CA  
Dallas, TX

### China



Hanghou, Shenzhen  
Hong Kong - Warehouse

### Morocco



Casablanca - Support Center, EMEA

### Venezuela



Venezuela- Support Center, LATM

### Netherlands



Moerdijk - Warehouse

## Awards & Recognition



14 time winner



9 time winner



2 time winner



## Global Presence



Over 10 Years of Growth and Innovation...

# VoIP Interoperability

## IP-PBX and Softswitch



## ITSPs



## Door Intercom



## VoIP Applications, Services, and Hardware





# Grandstream's Long History of Open Source

# SIP



# ANDROID

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**EMPATIQ**  
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# Full Unified Communication Solution



# VoIP Product Line

## Analog Telephone Adapter:



HT502



HT503



HT701



HT702/704

## Analog VoIP Gateways:



GXW400x



GXW410x



GXW42xx

## IP PBX Appliances:



UCM6102/4



UCM6108/16



UCM6510

## Small Business IP Phones:



GXP1610



GXP162x



GXP1628

## Enterprise IP Color Phones:



GXP2130



GXP2140 + EXT



GXP2160



GXP2200 EXT

## IP Multimedia Video Phones:



GXV3240 + EXT



GXV3275

## DECT IP Phones:



DP715



DP710

## Multimedia IP Phones

GXV3240 + EXT



- Android™ 4.2
- 4.3" capacitive touchscreen
- 6 lines, 5-way audio conference, and no-cost 3-way video conferencing
- 3rd party Android™ applications include Microsoft Lync, Skype, Twitter, Facebook
- Wi-Fi PoE, Bluetooth, USB, SD, mini HDMI, EHS
- Compatible with the GXP2200EXT extension module

GXV3275

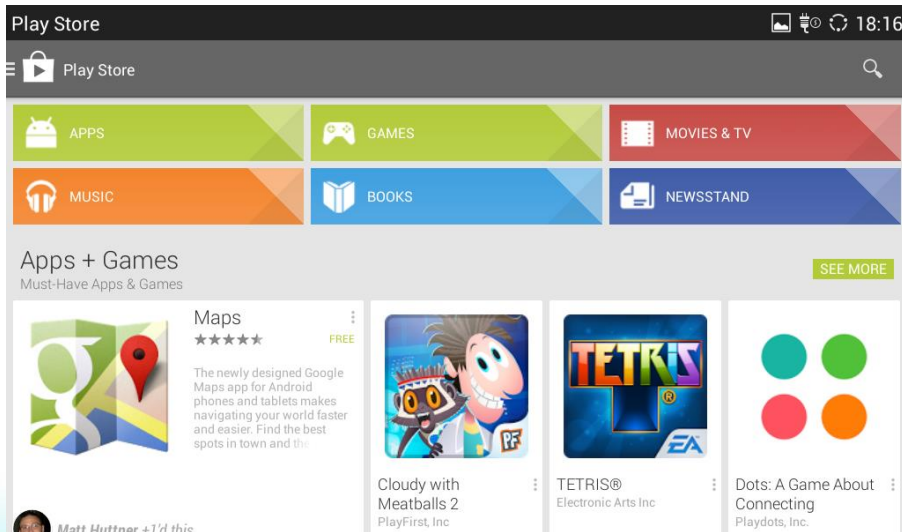


- Android™ 4.2
- 7" capacitive touchscreen
- 6 lines, 5-way audio conference and no cost 3-way video conference
- 3rd party Android™ applications include Microsoft Lync, Skype, Twitter, Facebook
- Wi-Fi PoE, Bluetooth, USB, SD, mini HDMI, EHS



# *Android + Video Conferencing + 6 Line IP Phone = GXV3200 series*





Download the hundreds of thousands of apps in the Google Play Store, including...

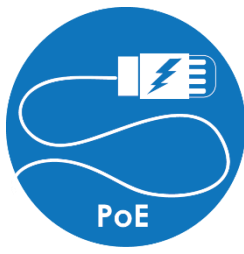
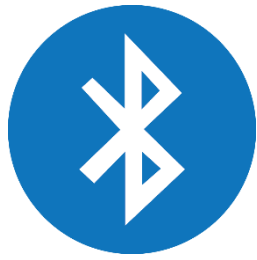


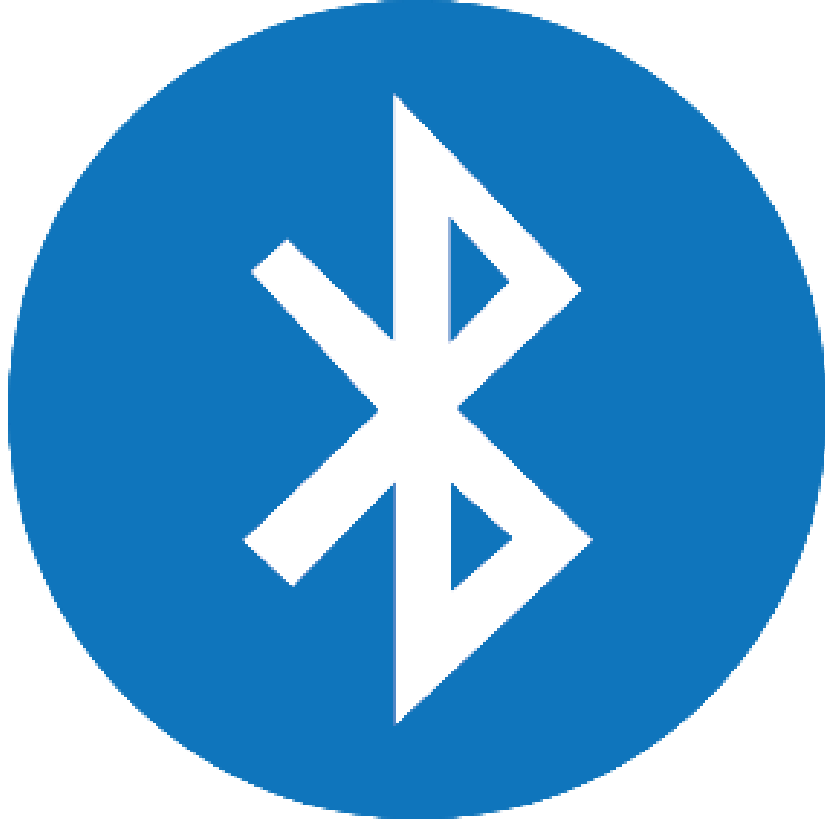
# Multi-Platform Video Conferencing









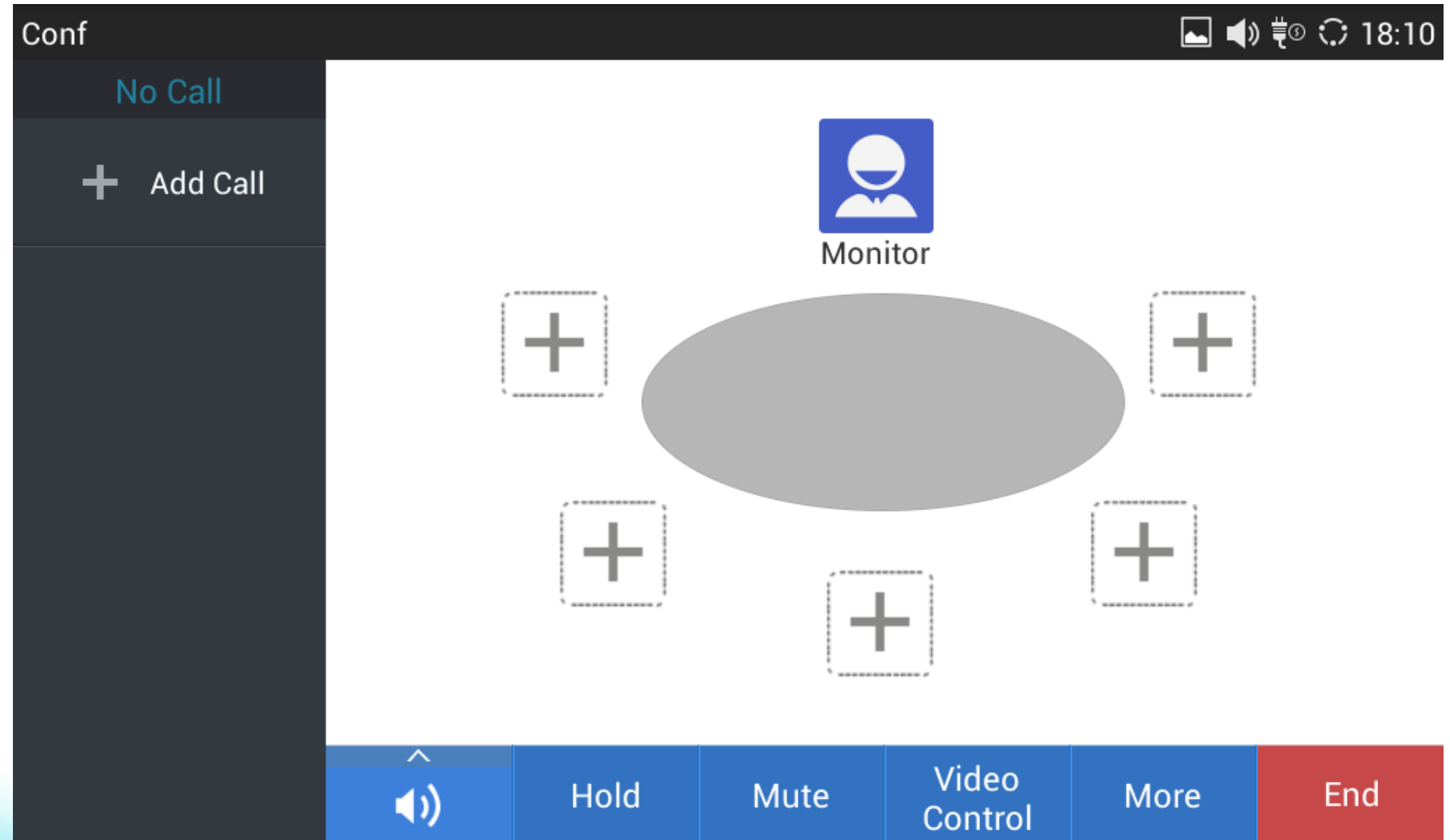


1. Call Transferring (calls incoming to paired mobile device)
2. Sync your mobile phone's contact book with GXV32xx



3. Use Bluetooth headsets

# 6-Way Voice Conferencing



# Integration with IP Video Surveillance Cameras





## Enterprise IP Phones

### GXP2130



- 2.8" color LCD
- BLF keys, 4 programmable keys
- 3 lines and 4-way conferencing
- Automated provisioning
- Dual Gigabit network ports
- Integrated PoE, EHS

### GXP2140 + EXT



- 480x272 backlit color LCD
- Integrated advanced Bluetooth, USB and EHS
- 4 SIP accounts, 5-way conferencing
- Dual Gigabit networks ports
- Up to 160 extensions with the GXP2200EXT module

### GXP2160



- 480x272 backlit color LCD
- Integrated advanced Bluetooth, USB and EHS
- 6 SIP accounts, 5-way conferencing
- 24 BLF dual-color extension keys
- Dual Gigabit networks ports

## SMB IP Phones

GXP1610



- 132 x 48 LCD screen
- 1 line, 2 call appearances, 3 XML programmable keys
- 3-way conferencing
- Integrated EHS

GXP1620/1625



- 132 x 48 backlit LCD screen
- 2 lines, 3XML programmable soft keys
- 3-way conferencing
- Integrated EHS
- PoE on the GXP1625 only

GXP1628



- 132 x 48 backlit LCD screen
- 2 lines, 3XML programmable soft keys, 8 BLF keys
- 3-way conferencing
- Integrated PoE and EHS
- Dual Gigabit ethernet ports

## Grandstream Analog Gateways

**GXW400x**



- 4 or 8 ports
- Multiple SIP profiles
- Auto provisioning
- Telephony features include 3-way conference, hold, transfer and hunt group

**GXW410x**



- 4 or 8 FXO ports
- Multiple SIP profiles
- Auto provisioning
- Interoperable with leading IP-PBXs, soft-switches and SIP-based environments

**GXW42xx**



- High density 16/24/32/48 port models
- Multiple SIP profiles
- Auto provisioning
- Gigabit network port
- Rack mountable

Coming Soon

# Video Conferencing System



## GVC3200

### Video Conferencing System

- Android 4.4 based, with embedded MCU
- Up to 4-way 1080p conference and remote screen viewing
- PTZ camera with 12x optical zoom
- Gigabit network port, Wi-Fi, Bluetooth, and Miracast
- 3 HDMI output ports, 1 VGA/HDMI input port, USB and SD



# Audio Conference Phone



## GAC2200

### Audio Conference Phone

- Android 4.4 based, up to 6-way conference
- 4.3" color LCD capacitive touch screen
- 3 microphones with 12ft voice pickup range, 1 8W speaker with 15ft coverage range
- Gigabit network port, integrated Bluetooth, USB and headset jacks

Introducing...

# UCM6100 & UCM6510 series IP PBX Appliances



UCM6102  
UCM6104



UCM6108  
UCM6116



UCM6510



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## UCM6100 series

IP PBX Appliance



- 2/4/8/16 FXO trunk port models, 2 FXS ports
- Up to 60 concurrent calls and up to 32 conference attendees
- Up to 500 SIP endpoints and up to 50 SIP trunk accounts
- Dual Gigabit network ports, integrated PoE, USB and SD ports
- Zero Configuration endpoint provisioning and no licensing fees



## UCM6510

IP PBX Appliance

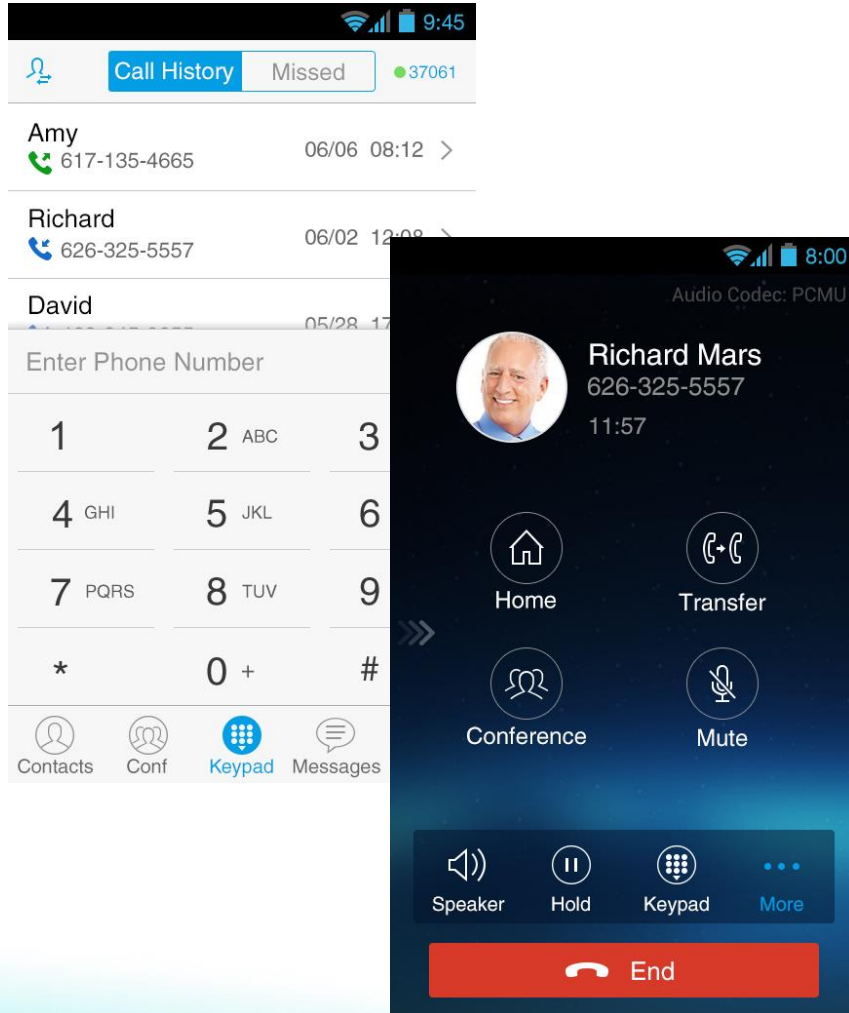


- E1/T1/J1 Interface
- 2 PSTN trunk FXO ports, 2 FXS ports with lifeline capability
- Up to 2000 SIP endpoints and up to 50 SIP trunk accounts
- Dual Gigabit network ports, integrated PoE, USB and SD ports
- Zero Config endpoint provisioning and no licensing fees



## Grandstream Wave

### Softphone Application For Android™



- Requires Android 4.0+
- Supports 6 SIP accounts, 6-way audio conferencing
- Features include call transfer, LDAP phonebook, virtual BLF keys and more
- Native integration with mobile devices including contacts, call history, and ringtones



# Grandstream IP PBX

## Voice

Secure, clear, and reliable

Conferencing

Comprehensive Codec Support

Endless Customizable Voice features

## Data

Phonebook files

CDR

Codec Transcoding

System Backup

Call Recording

Voicemail/fax to email



## Mobility

Softphone apps

Monitor your business

Connect Multiple Offices

## Video

Face to face video calls

Video Surveillance Integration

SIP Video

Video Codec Support

# General Specifications

- ◆ Up to 500 extensions
- ◆ # of FXO ports & concurrent calls:
  - 2 - 30 (UCM6102)
  - 4 - 45 (UCM6104)
  - 8 - 60 (UCM6108)
  - 16 - 60 (UCM6116)
- ◆ Gigabit ports with PoE Plus
- ◆ Each bridge supports up to 25 (UCM6102 & UCM6104) or 32 (UCM6108 & UCM6116) conference attendees
- ◆ Zero-configuration provisioning
- ◆ Simple setup/management with Web UI

A screenshot of the Grandstream PBX Status Web UI. The interface is in a dark blue theme. At the top, there's a navigation bar with "Status", "PBX", "Settings", and "Maintenance". The "Status" section is active, showing "PBX Status". On the left, there's a sidebar with "PBX Status", "System Status", and "CDR". The main content area displays several tables: "Trunks" with columns for Status, Trunks, Type, Username, and Port/Hostname/IP; "Extensions" with columns for Extension, Name/Label, Status, and Type; "Conference Rooms" with a table for 6300; "Interfaces Status" showing USB, LAN, FXS, and FXO ports; and "Parking Lot" with columns for Caller ID, Channel, Extension, and Timeout. The bottom of the page has a copyright notice: "Copyright © Grandstream Networks, Inc. 2013. All Rights Reserved."/>

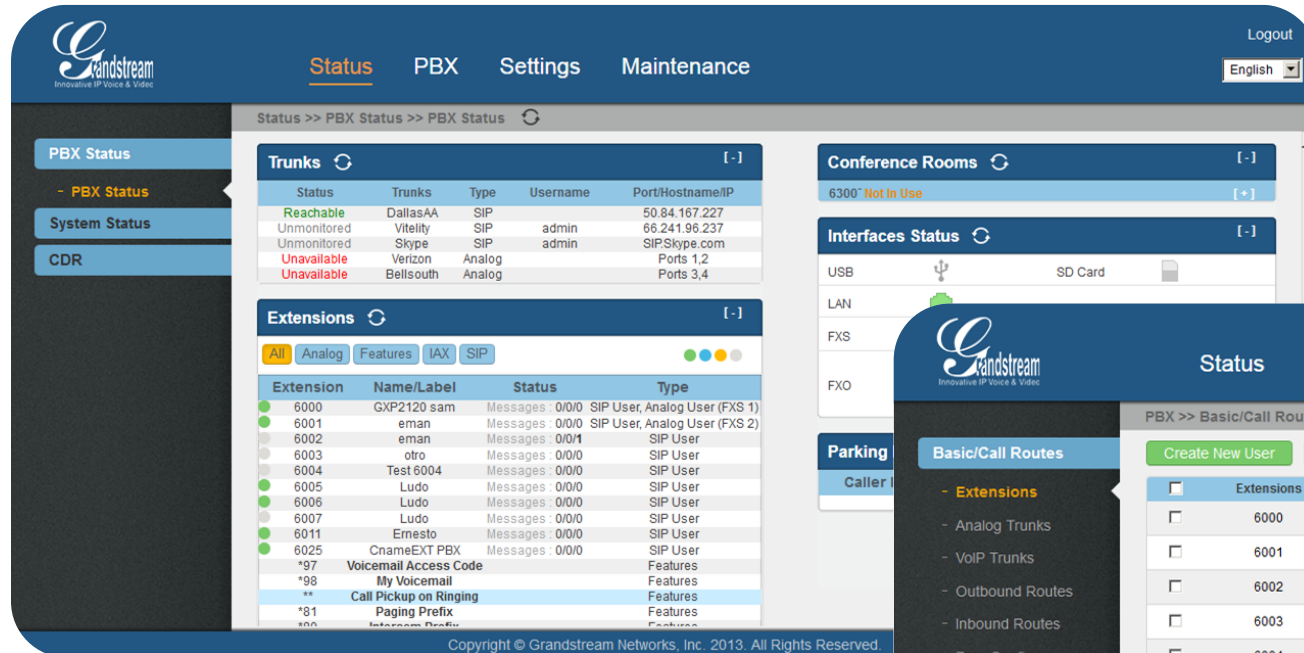
Status	Trunks	Type	Username	Port/Hostname/IP
Reachable	DallasAA	SIP		50.84.167.227
Unmonitored	Vitelity	SIP	admin	66.241.96.237
Unmonitored	Skype	SIP	admin	SIP:Skype.com
Unavailable	Verizon	Analog		Ports 1,2
Unavailable	Bellsouth	Analog		Ports 3,4

Extension	Name/Label	Status	Type
6000	GXP2120 sam	Messages: 0/0/0	SIP User, Analog User (FXS 1)
6001	eman	Messages: 0/0/0	SIP User, Analog User (FXS 2)
6002	eman	Messages: 0/0/1	SIP User
6003	otro	Messages: 0/0/0	SIP User
6004	Test 6004	Messages: 0/0/0	SIP User
6005	Ludo	Messages: 0/0/0	SIP User
6006	Ludo	Messages: 0/0/0	SIP User
6007	Ludo	Messages: 0/0/0	SIP User
6011	Ernesto	Messages: 0/0/0	SIP User
6025	CnameEXT PBX	Messages: 0/0/0	SIP User
*97	Voicemail Access Code		Features
*98	My Voicemail		Features
**	Call Pickup on Ringing		Features
*81	Paging Prefix		Features
6000	Internal Prefix		Features

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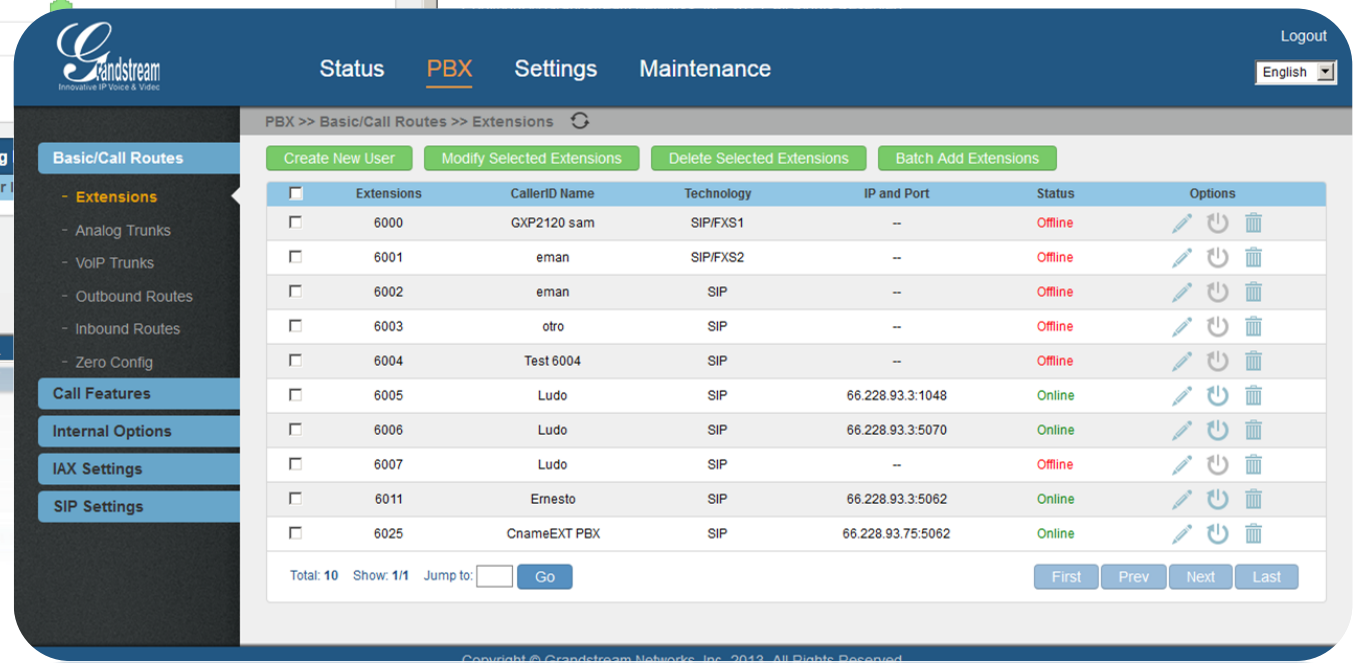
# Easiest Possible Setup...



This screenshot shows the 'Status' page of the Grandstream PBX interface. The left sidebar contains links for 'PBX Status', 'System Status', and 'CDR'. The main content area is divided into three sections: 'Trunks', 'Extensions', and 'Conference Rooms'. The 'Trunks' section shows a table with columns for Status, Trunks, Type, Username, and Port/Hostname/IP. The 'Extensions' section shows a table with columns for Extension, Name/Label, Status, and Type. The 'Conference Rooms' section shows a table with columns for Name, Type, and Port/Hostname/IP.

Status	Trunks	Type	Username	Port/Hostname/IP
Reachable	DallasAA	SIP	admin	50.84.167.227
Unmonitored	Vitelity	SIP	admin	66.241.96.237
Unmonitored	Skype	SIP	admin	SIP:Skype.com
Unavailable	Verizon	Analog		Ports 1,2
Unavailable	Bellsouth	Analog		Ports 3,4

Extension	Name/Label	Status	Type
6000	GXP2120 sam	Messages: 0/0/0	SIP User, Analog User (FXS 1)
6001	eman	Messages: 0/0/0	SIP User, Analog User (FXS 2)
6002	eman	Messages: 0/0/0	SIP User
6003	otro	Messages: 0/0/0	SIP User
6004	Test 6004	Messages: 0/0/0	SIP User
6005	Ludo	Messages: 0/0/0	SIP User
6006	Ludo	Messages: 0/0/0	SIP User
6007	Ludo	Messages: 0/0/0	SIP User
6011	Ernesto	Messages: 0/0/0	SIP User
6025	CnameEXT PBX	Messages: 0/0/0	SIP User
*97	Voicemail Access Code		Features
*98	My Voicemail		Features
**	Call Pickup on Ringing		Features
*81	Paging Prefix		Features



This screenshot shows the 'PBX >> Basic/Call Routes >> Extensions' page of the Grandstream PBX interface. The left sidebar contains links for 'Basic/Call Routes', 'Extensions', 'Analog Trunks', 'VoIP Trunks', 'Outbound Routes', 'Inbound Routes', 'Zero Config', 'Call Features', 'Internal Options', 'IAX Settings', and 'SIP Settings'. The main content area shows a table with columns for Extensions, CallerID Name, Technology, IP and Port, Status, and Options. The table lists various extensions and their configurations.

Extensions	CallerID Name	Technology	IP and Port	Status	Options
6000	GXP2120 sam	SIP/FXS1	--	Offline	
6001	eman	SIP/FXS2	--	Offline	
6002	eman	SIP	--	Offline	
6003	otro	SIP	--	Offline	
6004	Test 6004	SIP	--	Offline	
6005	Ludo	SIP	66.228.93.3:1048	Online	
6006	Ludo	SIP	66.228.93.3:5070	Online	
6007	Ludo	SIP	--	Offline	
6011	Ernesto	SIP	66.228.93.3:5062	Online	
6025	CnameEXT PBX	SIP	66.228.93.75:5062	Online	

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Thank you!

Questions?

