Grandstream DP715 IP DECT Base Station & Handset

Product Name: Grandstream DP715 IP DECT Base Station & Handset
Manufacturer: -
Model Number: DP715

Grandstream DP715 VoIP DECT Phone - Base Station and Handset

Please Note: The Grandstream DP715 has been discontinued. For an alternative, we recommend the Grandstream DP720.

The Grandstream DP715 is a VoIP DECT phone that is both powerful and affordable. The DP715 is simple to configure and of a high quality, making it ideally suited for both small business and residential users.

Grandstream DP715 Key Features

ï¿½ DECT base station registers up to 5 DECT handsets and up to 4 concurrent calls simultaneously
ï¿½ Advanced telephony features including Caller ID, Call Waiting, 3-Way Conference, Transfer, Forward, Do Not Disturb, Message Waiting Indication, auto answer, multi-language voice prompt, flexible dial plan
ï¿½ Secure and automated provisioning using HTTP/HTTPS/Telnet/TFTP, multiple SIP accounts, SIP over TCP/TLS, SRTP
ï¿½ Support comprehensive voice codecs including G.711, G.723.1, G.729A/B, G.726 and iLBC
ï¿½ When multiple handsets share the same SIP account, Hunting Group option may flexibly support:- Linear Mode, all phones ring sequentially in the predestinated order - Parallel Mode, all phones ring concurrently and after one phone answers the rest phones may place new calls - Shared Line Mode, all phones ring concurrently and always share the same line similar to an analog phone
ï¿½ Multi-Languages - English, German, French, Spanish, Dutch, Italian, Czech, Danish, Greek, Norwegian, Polish, Portuguese, Russian, Swedish, Turkish
ï¿½ Currently Pending - TR069, IPV6
ï¿½ The DP715 is NOT compatible with Skype

The compact size, superb voice quality, rich feature set, market leading price-performance and wide range radio coverage enable consumers to maximize the power of IP voice application and mobility for a minimum investment. DP715 is SIP and DECT compliant and field proven for flexible deployment.

ï¿½ DECT base station registers up to 5 DECT handsets and talks to up to 4 handsets concurrently
ï¿½ Advanced telephony features including Caller ID, Call Waiting, 3-Way Conference, Transfer, Forward, Do Not Disturb, Message Waiting Indication(Stutter Tone), auto answer, multi-language voice prompt, flexible dial plan
ï¿½ Secure and automated provisioning using HTTP/HTTPS/Telnet/TFTP, multiple SIP accounts, SIP over TCP/TLS, SRTP

Grandstream DP715 Technical Specification
Air Interfaces

ï¿½ Telephony standards: DECT / GAP Frequency range: 1880 - 1900 MHz (Europe), 1920 - 1930 MHz (US) Number of channels: 120 (Europe), 60 duplex (US) Modulation: GFSK Speech coding: 32 kbit/s Emission power: 10 mW (average power per channel) Range: up to 300 m

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outdoors, maximum of 50 m in buildings

Networking Interface

ï¿½ One 10/100Mbps auto-sensing Ethernet port (RJ45) (DP715 Base Station only)

LED Indicators

ï¿½ Base Station : Power, Network, Register, Call

Handset Display

ï¿½ 1.7" 102x80 FSTN LCD with color backlight

Factory Reset Button

ï¿½ Yes (DP715 Base Station only)

Audio Interface

ï¿½ Handsfree speaker (Handset only)

Voice over Packet Capabilities

ï¿½ Base Station : Dynamic Jitter Buffer Handset : Speakerphone with Acoustic Echo Cancellation

Voice Compression

ï¿½ G.711 with Annex I (PLC) and Annex II (VAD/CNG), G.723.1, G.726-32 AAL2, G.729A/B, iLBC

Telephony Features

ï¿½ Caller ID display or block, call waiting, Flash, blind or attended transfer, forward, hold, do not disturb, 3-way conference

QoS

ï¿½ Layer 2 (802.1Q VLAN/802.1p), Layer 3 (ToS, DiffServ, MPLS)

IP Transport

ï¿½ RTP/RTCP

DTMF Method

ï¿½ In-audio, RFC2833 and/or SIP Info

IP Signaling

ï¿½ SIP (RFC 3261)

Multiple SIP accounts per base station

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ï¿½ Up to five (5) distinct SIP accounts per system; Independent SIP account per handset; Multiple handsets per SIP account

Hunting Group

ï¿½ Linear mode; Parallel mode; Shared Line mode

Provisioning

ï¿½ HTTP, HTTPS, TELNET, TFTP, TR-069 (pending), secure and automated provisioning

Security

ï¿½ Security protection: SIP over TLS and SRTP.

Device Management

ï¿½ Web interface or secure (AES encrypted) central configuration file for mass deployment

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ï¿½ Web interface or secure (AES encrypted) central configuration file for mass deployment. Support device configuration via built-in IVR, Web browser or central configuration file through TFTP, HTTP or HTTPS. Auto/manual provisioning system. NAT-friendly remote software upgrade for deployed devices including behind firewall/NAT. Syslog support

Phonebook(Per Handset)

ï¿½ 200 numbers (up to 24 digits) with an associated name (up to 16 characters); 10 outgoing call entries; 30 incoming calls entries

Multi-language

ï¿½ Base Station Web UI: English; Voice Prompt : English, Spanish; Handset LCD Menu (15): English, French, German, Spanish, Dutch, Italian, Czech, Danish, Greek, Norwegian, Polish, Portuguese, Russian, Swedish, Turkish.

Multi-language Input

ï¿½ English; Latin; Greek; Russian

Polyphonic Ringtones

ï¿½ 18 different ringer melodies are available to indicate an incoming call (internal intercom or external VoIP)

Please Enquire