

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

- DECT base station registers up to 5 DECT handsets and up to 4 concurrent calls simultaneously
- 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
- Secure and automated provisioning using HTTP/HTTPS/Telnet/TFTP, multiple SIP accounts, SIP over TCP/TLS, SRTP
- Support comprehensive voice codecs including G.711, G.723.1, G.729A/B, G.726 and iLBC
- 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
- Multi-Languages - English, German, French, Spanish, Dutch, Italian, Czech, Danish, Greek, Norwegian, Polish, Portuguese, Russian, Swedish, Turkish
- Currently Pending - TR069, IPV6
- 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.

- DECT base station registers up to 5 DECT handsets and talks to up to 4 handsets concurrently
- 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
- Secure and automated provisioning using HTTP/HTTPS/Telnet/TFTP, multiple SIP accounts, SIP over TCP/TLS, SRTP

Grandstream DP715 Technical Specification

Air Interfaces

• 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

1/2 One 10/100Mbps auto-sensing Ethernet port (RJ45) (DP715 Base Station only)

LED Indicators

1/2 Base Station : Power, Network, Register, Call

Handset Display

1/2 1.7" 102x80 FSTN LCD with color backlight

Factory Reset Button

1/2 Yes (DP715 Base Station only)

Audio Interface

1/2 Handsfree speaker (Handset only)

Voice over Packet Capabilities

1/2 Base Station : Dynamic Jitter Buffer Handset : Speakerphone with Acoustic Echo Cancellation

Voice Compression

1/2 G.711 with Annex I (PLC) and Annex II (VAD/CNG), G.723.1, G.726-32 AAL2, G.729A/B, iLBC

Telephony Features

1/2 Caller ID display or block, call waiting, Flash, blind or attended transfer, forward, hold, do not disturb, 3-way conference

QoS

1/2 Layer 2 (802.1Q VLAN/802.1p), Layer 3 (ToS, DiffServ, MPLS)

IP Transport

1/2 RTP/RTCP

DTMF Method

1/2 In-audio, RFC2833 and/or SIP Info

IP Signaling

1/2 SIP (RFC 3261)

Multiple SIP accounts per base station

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

Hunting Group

• Linear mode; Parallel mode; Shared Line mode

Provisioning

• HTTP, HTTPS, TELNET, TFTP, TR-069 (pending), secure and automated provisioning

Security

• Security protection: SIP over TLS and SRTP.

Device Management

• Web interface or secure (AES encrypted) central configuration file for mass deployment

Device Management

• 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)

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

Multi-language

• 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

• English; Latin; Greek; Russian

Polyphonic Ringtones

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

Please Enquire
