IP Phone DBC 433 01

DESCRIPTION

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TRADEMARKS

1 DESCRIPTION

DBC 433 01 is a desktop IP terminal for business users. The IP communication protocol is H.323 supplemented by the WAP protocol for advanced telephony features.

Common standard voice codec's are supported including G.722 wide band codec.DBC 433 01 has a three line display (128 x 32 pixels).

The phone has two soft keys, two navigation keys, a standard key pad and twelve function keys. DBC433 01 has monitoring mode with the loudspeaker, which means that it is possible to listen with the loudspeaker but as soon as the user wants to speak, the handset must be used.

The phone has support for security with signal encryption according to TLS and media encryption according to SRTP.

2 HARDWARE

2.1 PRODUCT VERSION

Product Code	Product name	Color
DBC433 01/012	Mitel 7433	Body: Dark grey Display panel: Silver metallic grey

2.2 DIMENSIONS AND WEIGHT

Depth = 206 mm, width = 173 mm, height = 110 mm

Height value in max tilting position, width value in lowest tilt position

Total weight = 707 g

2.3 FAILURE RATE PER YEAR

Less than 2 % per year.

2.4 ACCESSORIES AND SPARE PARTS

Spare parts that users can replace themselves are:

- handset
- handset cord
- standard foot console.

Normally these phones are not repaired. Instead a swap repair procedure is used.

Accessories:

- Ethernet line cord.

Normally these phones are not repaired. Instead a swap repair procedure is used.

3 SOFTWARE

3.1 SOFTWARE FILES

The software to be upgraded in the phone must be stored on a web server with HTTP protocol. This web server is called SW server. The following files are stored on the SW server:

Application (CAA 158 0067). This file contains the application software for the phone.

Configuration file (CAA 158 0064). This file contains information about the version of the software to be used and other configuration data. The configuration file has to be adapted for each installation. The configuration file is common for all phones in the site or in the domain.

Language file (CAA 158 0063). The language file containing all the text strings for all the languages that are supported.

Super boot (CAA 158 0068) It keeps track of which one of the two application versions stored in the flash memory that is valid. The super boot starts the application.

When the phone is powered up, the phone fetches the configuration file from the software server. If the software version defined in the configuration file is different than the software version in the phone, the phone fetches the application file from the software server. The super boot software is updated if needed (it cannot be defined in the configuration file). The new software is automatically stored into the flash memory in the phone. The software of the phone can be upgraded and downgraded.

The latest software files can be downloaded from Astra Service Plaza.

3.2 SOFTWARE STRUCTURE

The application code is divided into the blocks: Voice Handler, H.323, UI, WAP, Web server and OMD.

The **Voice Handler** handles the DSP and controls the codecs. This block is responsible for the managing of voice channels, ring signals and tone generation.

The **H.323** block handles the call control. It follows the H.323 recommendation for ITU-T including RAS, H.225 and H.245. Note that services like transfer etc. are using a WAP protocol, see below. The codec settings and RTP information (address and port) are transferred to the Voice Handler block.

The **UI** block is the main task and is responsible for starting other tasks. It also controls the I/O devices of the IP phone, except the Ethernet driver. Examples of these devices are the keyboard, display and speaker. The UI has a state-machine which provides call control.

The **WA**P block provides the WAP protocol which allows service invocation and updating the display remotely from the gatekeeper.

The **Web serve**r block provides a way for the user, or a system administrator, to make changes in the phone book, the call log, shortcut keys or in some configuration parameters such as IP address or the Gatekeeper address etc.

The **OMD** (Operator Media Device) block provides call handling, registration etc. in a PBX operator solution.

4 TECHNICAL SPECIFICATIONS

4.1 FEATURE LIST

- All personal settings (including the call list) can be stored centrally in a XML file, and be retrieved at free seating.
- Possibility to have different ring signals for different calling persons.
- Log on/off function with three different modes
- Free seating by log on/off
- Call list with 50 entries
- Personal Contacts with 1000 entries. The application My Dialog Contacts gives possibility to merge the contacts from Microsoft® Outlook® Contacts.
- Branch office survivability support. The phone can register towards the branch office gatekeeper and re-register towards the main gatekeeper when it works again.
- Support for emergency call in log off state.
- Web server interface for operation and maintenance of the phone, by end user or system administrator.
- Interface to IP Phone Administrator which keeps track of the status of all the phones in the network.
- QoS report for the last 10 calls.
- Support for central Voice Recording.
- The phone can be used to handle the speech (OMD, Operator Media Device) in an Operator Work Station solution.
- Gatekeeper discovery with multi-cast signaling or by using the configuration file for the phone.
- Automatic software updates check every 24 hour (only for phones that are not registered).
- Possibility to reset the phone to factory default data.
- Translation of the display text strings to 19 languages.
- 2 navigation keys, 2 soft keys, 12 function keys and a standard key pad.
- Hearing aid and acoustic shock protection
- Display contrast and brightness can be changed in the phone.

4.2 IP INTERFACES

The phone supports the following protocols:

Standard	Function	Comment
DHCP	Used to obtain configuration information for operation in an IP network	IP addresses and vendor specific information

Standard	Function	Comment
DNS service records	Category of data in DNS specifying information on available services.	To retrieve IP addresses to the: 1. SW server 2. IP Phone administrator
HTTP	1. For downloading the phone firmware 2. Web interface for maintenance of the phone	
H.323v4	Umbrella recommendation for a number of protocols and media in a packet network	
H.225	1. Registration towards the gatekeeper. 2. Call signaling (a subset of Q.931)	H.225 is included in H.323
H.245	The capability exchange and logical channel management	H.245 is included in H.323
IEEE 802.1p	Mechanism for Quality of Services (QoS)	Diffserv is also supported
IEEE 802.1Q	VLAN tagging	
IEEE 802.1X	Standard for port-based network access control	Phone supports EAP-MD5
RFC3550	1.RTP: Real time protocol, defines media stream in a packet network 2.RTCP: RTP Control Protocol monitors the quality of the data distribution.	
SIP	Session Initiated Protocol	1.Used towards the voice recording equipment 2. Manufacturing tests
SRTP	Secure Real Time Protocol	Encryption of media
SSH	Secure Shell	Secure log on for maintenance of the phone
TLS	Transport Layer Security	Encryption of signalling
10/100 Mbit/s T/TX	Built-in 2-port Ethernet switch	

4.3 VOICE CODECS

The following voice codes are supported:

- G.722, wide band codec, is only used when direct media. (Cannot be used with MX-ONE TSW)
- G.711 A-law
- G.711 u-law
- G.729a
- G.729ab
- G.723.1

The codecs are by default chosen in the same order as in the list above, in the case when it is the phone and not the gatekeeper that decides which codec to use. This order can be changed via the phone configuration file. It is possible to change the packet size for:

- G.711: possible values are 10,20 or 30 ms. The default value is 30 ms.
- G.729a and G729ab: possible values are 10,20,30,40, 50 or 60 ms. The default value is 30 ms

4.4 VIOCE TRANSMISSION

Expressions used in this section:

- Receiving Loudness Rating, RLR
- Sending Loudness Rating, SLR

Transmission and receiving loudness rating are according to ETSI requirements:

- Handset: ETS 300 245-2, TIA 810B and TIA920

4.4.1 HANDSET MODE

Function	Value dB	Tolerance +/- dB	Comment
SLR	+7	3.5	
RLR	+3	3.5	
Side tone, STMR	+13 to +18		Volume control adjustable from 12 dB amplification to 9 dB attenuation, relative to the nominal value
Microphone			Electret
Receiver			Dynamic

4.4.2 HEADSET MODE

There is no headset support.

4.4.3 HANDSFREE MODE

There is no handsfree mode, only monitoring mode.

4.5 CALL PROGRESS TONES

The default tones are according to Euro-ISDN.

Type of tone	Frequency	Comment
Dial tone	425 Hz	Continuous
Special dial tone	500 ms on, 50 ms off, 425 Hz	Repetitive
Busy tone	500 ms on, 500 ms off, 425 Hz	Repetitive
Alerting tone	1000 ms on, 4000 ms off, 425 Hz	Repetitive
Congestion tone	200 ms on, 200 ms off, 425 Hz	Repetitive
Special information tone	333 ms on at 950 Hz, 0 ms off, 333 ms at 1400 Hz, 0 ms off, 333 ms at 1800 Hz, 0 ms off, 1000 ms off	Repetitive
Call waiting tone	200 ms on, 600 ms off, 200 ms on, 3000 ms off at 425 Hz	Repetitive
On hold tone	10 ms on, 150 ms off, 10 ms on 5000 off at 425 Hz, 300 Hz.	Repetitive

Level: -10dBm0

The tones can be adapted to market specific values via the configuration file.

4.6 LED CADENCES

Cadence	Time	Comment
Continuous	425 Hz	A function is activated (mute, speaker etc.)
Slow blinking	500 ms on, 500 ms off	Parked call
Fast blinking	300 ms on, 300 ms off	Incoming call
Lit with fast blink	1850 ms on, 50 ms off, 50 ms on, 50 ms off.	Active call

These are the default values, which can be changed via the configuration file.

4.7 TONE RINGER

4.7.1 TONE RINGER CADENCES

Type of ring signal	Time	Comment
Internal call	1 s ringing, 5 s silence	Repetitive
External call	350 ms ringing, 300 ms silence, 350 ms ringing, 5000 ms silence	Repetitive
Call back	300 ms ringing, 400 ms silence	Repetitive
Extra	350 ms ringing, 300 ms silence	Repetitive

These are the default values, which can be changed via the configuration file.

4.7.2 RING SIGNALS

These are 10 different ring signals.

4.7.3 ALERT SIGNALS

The following types of alert signals are available:

- Periodic (repetitive). This is the default value.
- Delayed periodic. The delay time can be set via the configuration file.
- One muted. Only one ring signal with low volume.
- Delayed one muted. The delay time can be set via the configuration file.
- Visual only

The types of alert signals above can be set independently for:

- The first incoming call (normally on line 1)
- The second and third incoming call (normally on line 2 and line 3
- Monitor keys (MNS)

4.8 POWER

Option	Specification	Idle	Max
Power over Ethernet	IEEE 802.3af power class =1	less than or equal to 2,0 W	less than or equal to 2,6 W
AC/DC Adapter, 5V	RES 141 319/1 (EU) RES 141 319/2 (US) RES 141 319/3 (UK)	less than or equal to 1,4 W	less than or equal to 1,8 W
	Only RES 141 319 without phone		less than or equal to 0,3W

The power consumption values are without any PC connected to the PC outlet in phone.

4.9 CLIMATIC ENVIRONMENT RANGES

- Operating temperature +5° - +45°C
- Storage temperature +/-0°- +70°C
- Relative humidity: 5 - 90%

4.10 REGULATORY COMPLIANCE

- Acoustic chock protection: ETS300 245-2, EN71-1
- Hearing Aid Compatibility (HAC): ITU-T P.370 (8/96) and FCC; Part 68.316 subpart D American.
- EU WEEE directive
- EU RoHS directive 2002/95/EC
- Chinese ROHS directive

For the declaration of conformity, see www.mitel.com/sdoc.