# Configuration File for DBC 43X and DBC 44X

DESCRIPTION

# 1 GENERAL

This description of the configuration file is valid for the following IP telephones:

- DBC 433 (Mitel 7433)
- DBC 434 (Mitel 7434)
- DBC 444 (Mitel 7444)
- DBC 446 (Mitel 7446)

When the IP telephones are used with MX-ONE 3 (or later) it is recommended that **MX-ONE Service Node Manager** is used. It has a graphical user interface support for the parameter settings in the configuration file for the telephones, see 2 Configuration file in a MiVoice MX-ONE Service Node environment on page 4.

The DBC 446 telephones support the H.323 protocol.

DBC 433, DBC 434 and DBC 444 can use the H.323 protocol.

The IP telephones use a configuration file to initiate parameters in the telephone. Examples of parameters that are possible to set in the configuration file are:

- which system the IP telephone shall register towards
- the version of the software to be used
- priority of the codecs
- assignment of function keys

The file shall be stored on a software server (web server) and the IP telephone uses the http protocol (with port number 80) to read the file. The other software files to be loaded into the IP telephone shall also be stored on the same software server.

The name of the configuration file must be:

- **d44x01-config.txt** for DBC 444 and DBC 446 (product number CAA 158 0058)
- **d43x01-config.txt** for DBC 433 and DBC 434 (product number CAA 158 0064)

To detect faults in the configuration file after it is downloaded to the phone, initiate a self test from the phone by pressing , *****, and **4** simultaneously for at least one second.

The configuration file can be adapted or changed by the system administrator with any text editor.

The configuration file is read when the telephone is:

- rebooted, either by an order from the PBX or manually or through the web interface.
- not registered and when entering the administrator mode (press C,*,5)
- not registered, then the telephone will fetch the configuration file from the SW-server every 24:th hour.

## 1.1 REFERENCES

For more information about the software server and about the directory structure where to store the configuration file:

- See *Installation instructions for DBC 43x and DBC 44x*.

# 2 CONFIGURATION FILE IN A MIVOICE MX-ONE SERVICE NODE ENVIRONMENT

When using IP phones in an environment with MX-ONE 3.0 or later, it is recommended that **MX-ONE Service Node Manager** is used when working with the configuration files for IP phones. The information regarding data identifiers (parameters) in this document is also available in the online help for MX-ONE Service Node Manager.

The software needed for this tool, see Figure 1.

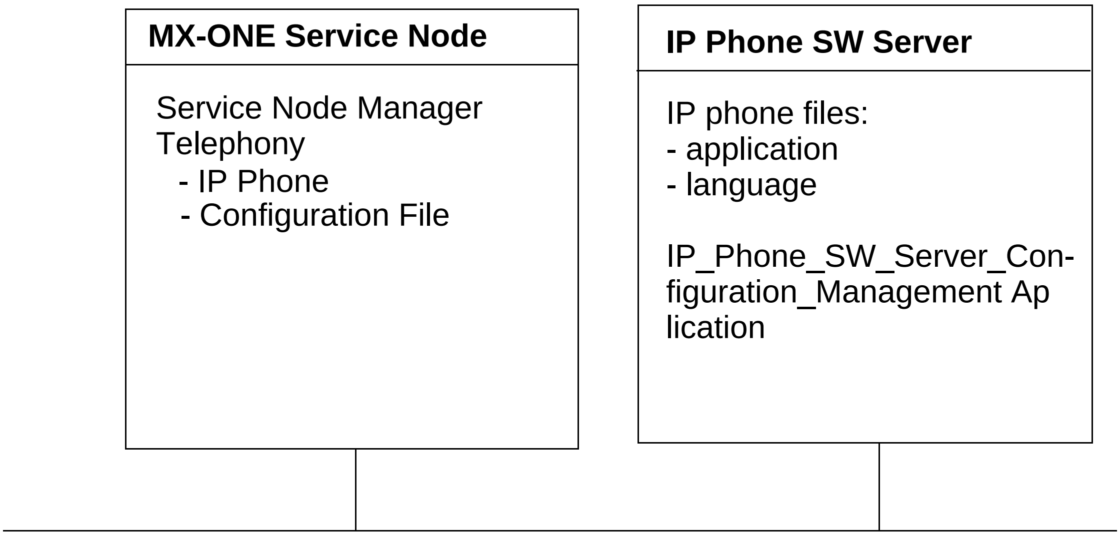



**Figure 1: The deployment of Service Node Manager, IP Phone Configuration File**

## 2.1 CREATE A CONFIGURATION FILE - NEW INSTALLATION

The procedure at new installation is:

- Installation of IPPhone SW Server.
- Create a configure file for DBC 43X and DBC 44X.

The software to be used in the IP phone software server can be downloaded from the Service Support Plaza. The software consists of two parts:

- IP phone files: application and language files. If the super boot, key panel unit (KPU) firmware or display panel unit (DPU) firmware have to be updated, these files are also needed.
- IP Phone SW Server Wizard. This package includes also the Tomcat web server.

### 2.1.1 INSTALLATION OF IP PHONE SW SERVER

See the IP Phone SW Server installation instruction in document *IP PHONE SOFTWARE SERVER.*

### 2.1.2 REGISTER AN IP PHONE SERVER FOR DBC 44X OR DBC 43X

**Do as follows:**

1. **For DBC 44X**: create the **dbc44x01** directory under the web-server root: **jakarta-tomcat-4.1.31\webapps\ROOT** on the IP Phone SW server (where -4.1.31 is an example).
   or
   **For DBC 43X:** create the **dbc43x01** directory under the web-server root: **jakarta-tomcat-4.1.31\webapps\ROOT** on the IP Phone SW server (where -4.1.31 is an example).
2. **For DBC 44X:** Copy the IP phone files (application and language) for DBC444 and DBC446 to the **dbc44x01** directory. Manager SN will store the created configuration file d44x01-config.txt under this directory.
   or
   **For DBC 43X:** Copy the IP phone files (application and language) for DBC43x to the **dbc43x01** directory. Manager SN will store the created configuration file d43x01-config.txt under this directory.
3. Log in to **MX-ONE Service Node Manager (SNM)**. The welcome page is displayed.
4. Select **Telephony**, **IP Phone** and then **SW Server**. The page **IP Phone SW Server** is displayed.
5. Click **Add**. The page **IP Phone SW Server - Add** is displayed.
6. Register the **IP Phone SW Server;** type information in the fields **Server name**, **IP Address** and **Port Number**.
7. Click **Apply**. If the operation went well, you will get the information, **Add operation successful for XX:** (XX=name of the server).

### 2.1.3 CREATE A CONFIGURE FILE

**Do as follows:**

1. Select **Telephony**, **IP Phone** and then **Configuration file** on the **Welcome page** in **SNM**. The page **IP Phone Configuration file** is displayed.
2. Click **Add**, to open a new configuration file. The **IP Phone Configuration File - Add step 1/10** is dispaplayed.
3. Type configuration data in the 10 steps. If you need help, use the **SNM online** help to the right.
4. Click **Apply** to finish the steps and the configuration, and to store data under the correct directory in the IP Phone Software Server.

**To force the telephones to fetch the new configuration file there are a number of cases:**

- If the telephones are not started yet: connect the power and the telephones will fetch the new configuration file.
- If the telephones are already registered to the PBX, select the **Unregistration** option to force the telephones to fetch the new configuration file.
- If the telephones are started but not registered to the PBX:

– use the task IP Phone Administrator (in MX-ONE Service Node Manager) to log on to the telephones and initiate restart from the administrator Web interface.

– the telephones will after less than 24 hours automatically fetch the new configuration file and if necessary download a new firmware.

– restart the telephones manually. For information on how to restart phones, see *Installation instructions for DBC 43x and DBC 44x.*

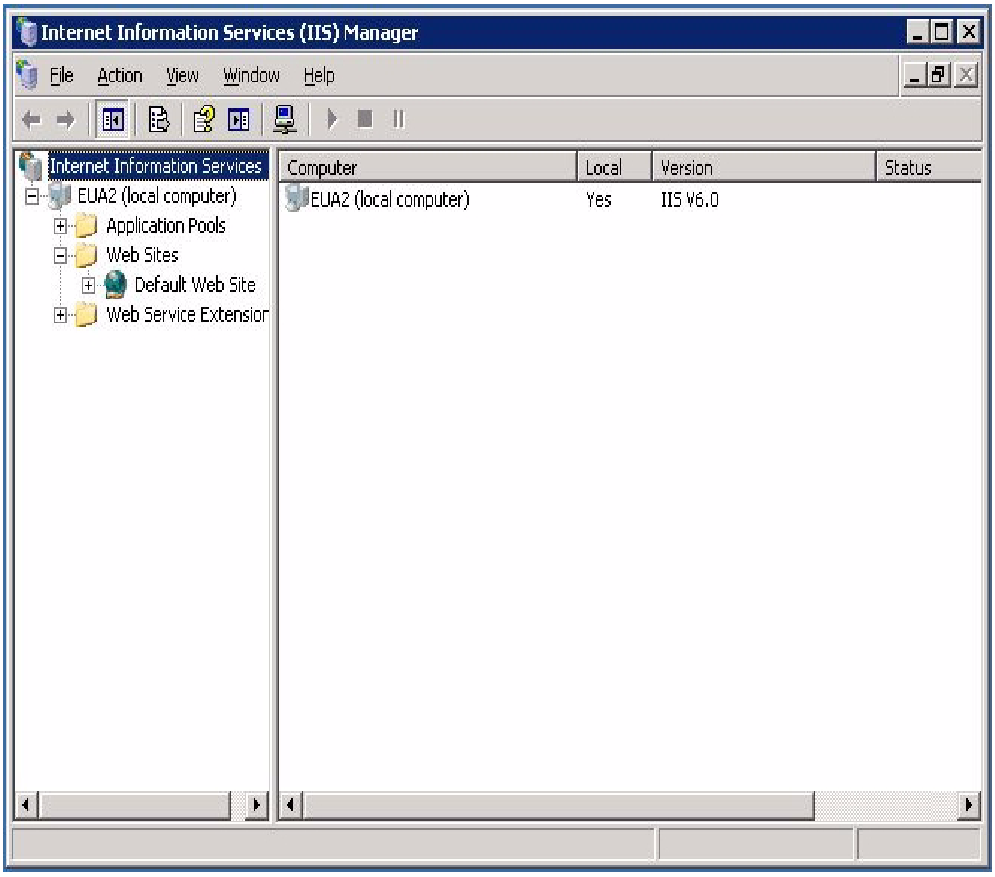
## 2.1.4 PORT CONFLICT

The *IP Phone SW Server Configuration Management Application* runs on a Windows server that often is used also by other applications and their own web servers. If a Windows IIS web server exists there can be a problem with a conflict between the ports for the web servers since both the Tomcat and IIS should use port 80. The IP phones must use port 80 for fetching the firmware files.

**Measures to avoid port conflict**

- Deploy IIS and run it on port 80
- Deploy IP Phone Configuration Management Application with Tomcat and run it on port 82
- Connect SNM to *IP Phone SW Server Configuration Management Application* on port 82.
- Set up the folder structure required by the terminals in Tomcat and copy application and language files into it. But not the configuration file. Example:
  C:\jakarta-tomcat-4.1.31\webapps\ROOT\d43x01\d43x0-applic_R1F.dat
  C:\jakarta-tomcat-4.1.31\webapps\ROOT\d43x01\d43x01-lang_R1E.txt
- Create the configuration file in SNM and store it in IP Phone Configuration Management Application
- Redirect IIS to tomcat for the terminal requests like this:

– Open C:\WINDOWS\system32\inetsrv\inetmgr.exe, navigate to Default Web Site




– Right click on Default Web Site and select New Virtual Directory. A wizard will start

– Enter the directory name to where the telephone firmware shall be stored as Alias, example: Mitel67xxi

– Enter the path to the family n folder under Tomcat, example: C:\jakarta-tomcat-4.1.31\webapps\ROOT\d43x01\d43x0-applic_R1F.dat

– Enable the Read option and finish the wizard

– You can now access the Tomcat folder with terminal settings on both port 80 as well as 82, while SNM can update the configuration file on port 82

– If subnets or telephony domains are defined for the configuration file in SNM, the path under Tomcat will include the subnet/telephony domain in its path. Update the IIS virtual directory link accordingly

# 3 DESCRIPTION

## 3.1 SYNTAX

The configuration file contains five different types of data:

- *Headers*, are used to create different groups of data identifiers: for example, [System], [Software], [Language], [WAP] etc.
- *Data identifier*, is a reserved word which ends with an equal-sign: for example, System=.
- *Dataless identifier,* is an identifier without data and is not followed by an equal-sign: for example, G.729A.
- *Data*, is the value after the equal-sign until the end of the line: for example, MD110
- *Comment*, a line starting with a semi-colon and the text until end of the line: for example, ; Default language is English

If a data identifier is enabled, the corresponding header must also be enabled.

The following example shows a combination that is **not allowed**:

; [StoreUserData]

EnableStoring=YES

## 3.2 SYSTEM

### 3.2.1 H.323 MODE

The header [System] has the following data identifiers:

- **System**. Which system (gatekeeper) the telephone shall register toward. The allowed values are:
  - **MD110**. To be used with ASB 501 04 (MD110 and MX-ONE Telephony Switch). The identifier gives the following features:

    – Function keys for line1, line2, inquiry, transfer, message, follow me, call back and free on 2:nd line. For more information, see 3.18 Shortcut Keys on page 22.

    – Uses Q.931 overlap sending, if overlap is set to **YES**.

    – WAP signaling and this enables the use of soft-keys.

    – Time is set via the WAP signaling or via an NTP server.
  - **MX-ONE**. To be used with MX-ONE Service Node. The identifier gives the following features:

    – Function keys for line1, line2, inquiry, transfer, message, follow me, call back and free on 2:nd line. For more information, see 3.18 Shortcut Keys on page 22.

    – Uses Q.931 overlap sending, if overlap is set to **YES**.

    – WAP signaling and this enables the use of soft-keys.

– Time and date are received from the NTP server, the primary gatekeeper, or the secondary gatekeeper.

– **BP-R16**. With the following features:

– Function key for line1.

– Name and display information received by the telephone in WAP messages and not in H.323 messages. Number is still received in H.323 messages.

– At off hook the speech channel (H.245) is connected and the dial tone generated by the gatekeeper is presented transparently by the telephone. Dial tone is generated by the gatekeeper after hook flash.

– Hook flash function.

– Message waiting and display text with use of WAP.

– Time and date are set via the WAP signaling.

– **MD-E**. With the following features:

– Function keys for access1, access2, inquiry, transfer and message indication. It is possible to receive a second call even when the free on second key is not used. It is possible to add digits after the TNS key is pressed.

– WAP signaling and this enables the use of soft keys and the PABX services.

– Uses Q.931 overlap sending, if overlap is set to **YES**.

– Time and date are set via the WAP signaling.

– **System data identifier omitted**. Has the following features:

– Line1 key

– At off hook, **setup** is sent to the gatekeeper and the digits are sent according to Q.931 (if overlap sending is used).

– Local dial tone is generated at off hook.

- **OverLap**. How the entered digits are sent to the gatekeeper, with the allowed values:
  - **YES**. The entered digits are sent one by one to the gatekeeper. The default value.
  - **NO**. All entered digits are sent in a block to the gatekeeper and the call is initiated by pressing the call key.
- **AutodialTimeout.** 1-20 seconds. The default time is 5 seconds. The time before the telephone sends the entered number to the PBX. The timout is only valid in en-block sending (OverLap=NO). If no timeout is wanted, disable the parameter as a comment.
- **RRQTtl**. *Registration Request Time to live* is used to check if the connection between the IP telephone and the gatekeeper is up. This parameter defines the length of the interval between time to live re-registrations in seconds. The parameter values can be:
  - The default value is 600 i.e. 10 minutes
  - The minimum value is 40 seconds.
  - **OFF**: *Registration Request Time to live* is not sent.

– 0 (zero): the default time will be used and the time to live field will not be included in Registration Request message.

The RRQTtl parameter is only used when the time to live value is not received (in the RCF message) from the gatekeeper.

The parameter is also used in the branch office scenario, to discover when the telephone shall register towards the backup gatekeeper and also to discover when the main site is up again and the telephone shall register towards the main site.

- **GatekeeperDiscovery**. Gatekeeper discovery is a method to find a gatekeeper, to which the IP telephone should register. This data identifier has two possible values:

  – **YES**. Automatic gatekeeper discovery will be used by default, but it can be disabled in the settings menu on the IP telephone

  – **NO**. Automatic gatekeeper discovery will not be used unless gatekeeper discovery is set to yes in the menu of the IP telephone.

  If the **GatekeeperDiscovery**data identifier is omitted, the default value **YES** will be used.

- **GatekeperID**. This is the gatekeeper identity of the PABX that the IP telephone should be registered to. This is used in the gatekeeper discovery procedure to find out which of the gatekeepers the IP telephone should register to, for MX-ONE, see operational directions for *IP EXTENSION*. It is possible to use wild cards, example: LIM* accepts all gatekeeper identities that begin with LIM.

- **Domain**. This is the telephony domain name to which the IP telephone is connected and it is used for **Gatekeeper discovery**, for MX-ONE see operational directions for *IP EXTENSION, IP*. This identifier is a text string. This domain name is a text string and only used when a domain name is not received from the DHCP server.

  If the **Domain** identifier is omitted in the configuration file and is not received from the DHCP server, no domain name will be used by the IP telephone.

- **PrimaryGKAddress.** This is the IP address to the primary gatekeeper, which will be used only if the **GatekeeperDiscovery** identifier is set to **NO** and gatekeeper discovery is set to the value default in the menu of the IP telephone. If this identifier is omitted and **GatekeeperDiscovery** is set to **NO** the IP telephone will use the manually set address in the **Settings menu** in the display.

- **SecondaryGKAddress.** This is the IP address to the secondary gatekeeper, which will be used if the primary fails. This identifier should only be used if **PrimaryGKAddress** is also used. Primary and secondary gatekeeper must be the same **system**, see heading [System].

- **AdminPassword.** This identifier is used to set the **Administrator password** which is used in SSH and in the built-in web server. The password must be in encrypted form, this is done in SSH by calling the function **enctryptPasswd "<password>"**, with the desired password as argument. The administrator user name is admin, and the name is fixed.

- **LogOffRestriction.** When all the telephones shall have the same log off restriction option, this parameter can be used. When this parameter is used in the configuration file, *it is not possible to change this parameter locally in the telephone*. The values are:

  – **LogOffAllowed**. The end-user is allowed to log on and log off the terminal. Se also parameter **LogOffTime** below.

- **DefaultNumberUsed**. The telephones are always logged on with a default number.
- **PermitIndividualLogOn**. The telephones are always logged on with their default number, but the end-user can log on with his/her individual number.

For more information see *Installation Instructions for DBC 43x and DBC 44x*.

- **LogOffTime=hh:mm.** where hh mean hours and mm minutes. When this parameter is enabled all the users will be logged off at the specified time. This parameter is only used when *LogOffRestriction=LogOffAllowed*. The default value is that this parameter is disabled.
- **OperatorNumber**. (DBC 433, DBC 434 and DBC 444 only). This parameter is used to set the common PBX operator number. This operator number is used when the end user wants to call the PBX operator using the telephone menu (that is, without dialing the number manually). There can be 1-5 digits in the operator number.

## 3.3 SOFTWARE

The header [Software] has the following data identifiers:

- **ApplicationFile**. The path and file name for the application for DBC 43x and for DBC 446. The path must be according to the description in the installation instructions, see section 2.1 References.
- **ApplicationVersion**. The version of the application software
- **DBC 444ApplicationFile**. The path and file name for the application for DBC444.
- **DBC 444ApplicationVersion**. The version of the application software for DBC 444.

If **ApplicationVersion** in the config file is not equal to the revision in the application software, shown in the display by pressing ,*, and *4*, the application file is always fetched from the Web server. Example: if the **ApplicationVersion** is R2G and the revision shown in the display in the telephone is R2F, the file with the application R2G will always be fetched.

## 3.4 802.1X

The header [802.1x] is used to setup the parameters for LAN access control according to IEEE802.1x.

Enabled data identifiers in the configuration file override the values in the phone when the configuration file is loaded.

The telephone has support for the protocols EAP-MD5 and EAP-TLS. In the later case certificates have to be loaded into the telephone.

The following identifiers exist:

- **LANAccessControl:** To enable or disable the IEEE802.1x function:
  - **AUTO**. The telephone will initiate IEEE802.1x signaling in the start up sequence and if the LAN supports IEEE802.1x, the telephone enables this function. This is the default value.

  - **NO**. The telephone disables the IEEE802.1x function.

- **StoreUserIDPassword: YES** or **NO**. If the user identity and the password shall be stored in the telephone or not. The default value is **YES**. This parameter is only valid when EAP-MD5 is used.
- **UserType**. Indicates if the user identity and password shall be valid for the telephone or for the end user. This parameter is only valid when EAP-MD5 is used.
  - **User**: The telephone will be logged off from the LAN when the end-user registers with a different extension number towards the PBX. The end user has to enter the LAN access control user identity and password each time he/she registers with a different extension number.
  - **Phone**: The telephone will not be disconnected from the LAN when entering a different extension number to register towards the PBX. The default value is **Phone**.
- **UserIdentity**: The user identity is loaded into the telephone from the configuration file.
- **UserPassword**: The user password is loaded into the telephone from the configuration file.

  The UserIdentity and UserPassword identifiers are only used when the MD5 protocol is used. These parameters are used in the case when all the telephones shall have the same user identity and password. This is a convenient option compared to enter this from the key pad, especially if many telephones shall be installed. The recommendation is to use a configuration file with the UserIdentity / UserPassword activated at installation and then remove these identifiers once the telephones are installed.
- **EAP_TLS**: Parameter defining which security protocol to use:
  - **ENABLED**. TLS is enabled. A certificate is mandatory.
  - **DISABLED**. MD5 is enabled. No certificate is used. This is the default value.
- **RootCert = file name.pem.** The file name of the root CA certificate stored on the software server. The path where the file shall be stored must be according to the Installation Instructions for DBC43x and DBC 44x. This parameter is only used when EAP_TLS is enabled.
- **ClientCert = file name.pem.** The file name of the client certificate stored on the software server. This parameter is only used when EAP_TLS is enabled.
- **ClientKey = clientkey.pem.** The name of file where the private key is stored on the software server. This parameter is only used when EAP_TLS is enabled.
- **ClientKeyPasswd = password.** The password for the private key of the client certificate. This parameter is only used when EAP_TLS is enabled.
- **RandomFile = file name.bin.** A random number used in the cipher algorithm. This file can be generated by the Radius server. This parameter is only used when EAP_TLS is enabled.

## 3.5 ABSENCE SERVICES

This is a group of procedures to be sent to the gatekeeper when absence services is used. Each procedure is a combination of digits, *, and #.

[Absence] is the header used for setting up these procedures. The identifiers are:

- **Profile** = code for activating a profile, code for deactivating profile. Example: *10*profileNo#,#10#. Only the digits can be changed.
- **FollowMe** = code for activating follow me, code for deactivating follow me.
- **ExtFollowMe** = code for activating external follow me, code for deactivating external follow me.
- **AbsenceReason** = code for activating absence reasons, code for deactivating absence reason. Example: *23*reasonCode#,#23#.

  The format of reasonCode in the above example is name of *reason=reason number*hhmm*, where hh is hours, mm minutes when a time is used, mmdd are used for dates where mm is month and dd is day.

  The star (*) between reason number and the time shall not be used in Business-Phone.

  If any of the codes for the AbsenceReason are within the comments (;) in the config file, it will not be displayed on the telephone. The total number of possible absence reasons are according to the list, but maximum 10 reasons can be used in a site:

  – **Lunch=0*hhmm, DefaultTime=hh:mm**
  – **Busy=1**
  – **Absent=2*hhmm, DefaultTime=hh:mm**
  – **Meeting=3*hhmm, DefaultTime=hh:mm**
  – **Trip=4*mmdd, DefaultTime=**
  – **Course=5*mmdd, DefaultTime=**
  – **Vacation=6*mmdd, DefaultTime=**
  – **DayOff=7*mmdd, DefaultTime=**
  – **GoneHome=8*mmdd, DefaultTime=**
  – **Illness=9*mmdd, DefaultTime=**
  – **OfficialMatter=n*hhmm, DefaultTime=hh:mm**
  – **FreeTime=n*mmdd, DefaultTime=**
  – **Special=n*mmdd, DefaultTime=**
  – **ParentalLeave=9*mmdd, DefaultTime=**

  When the user sets the absence reason, the telephone proposes a default return time. It is possible to change the default time by the parameter DefaultTime in the following way:

  – **DefaultTime=hh:mm**. This default return time will be presented in the menu *My Presence*. The parameter is useful when the default time shall be synchronized with the interception service system, in the case when this system proposes a default return time. Example: in the configuration file *Lunch=0*hhmm, DefaultTime=00:40* is set. The user selects lunch at 12:00. In the menu *My Presence*, the return time is presented as 12:40.
  – **DefaultTime=** without any parameter value. No default return time or default return date will be presented in the menu *My Presence*.
  – **DefaultTime.** The parameter is not defined. The default return time of 1 hour or the date of today as default return date will be presented in the menu *My Presence*. This is the default value.

For ASB 501 04 and MX-ONE: The date format must correspond to PARNUM=62 in the ASPAC command.

It is possible to change the Absence reason text string shown in the display by editing the language file, see description for *LANGUAGE FILE FOR DBC 43x*, *LANGUAGE FILE FOR DBC 444 and LANGUAGE FILE FOR DBC 446*.

## 3.6 AUTHORIZATION

The header [Authorization] is used to replace the digits with a dash (-) in the display for password codes and similar. The same identifier is used for all codes.

- **Code**= A function code after which entered digits are hidden by dashes (-). Examples:
  - **Code=*75*** means that all digits entered after *75* are displayed as dashes. If a user enters *75*1234*67609# (where, for example, 1234 is a password or PIN code and 67609 is a directory number), the display will show *75*--------.
  - To display digits as entered, the **n** switch is used. For example, **Code=*75*-*n#** means that digits entered between *75* and *n# are replaced by dashes, while digits between * and # are shown as entered. If a user enters *75*1234*67609# (where, for example, 1234 is a password or PIN code and 67609 is a directory number), the display will show *75*----*67609#.

## 3.7 AUTO NEGOTIATION

The header [AutoNeg] is used to setup speed and duplex mode for the LAN and PC port. The following identifiers are used:

- **AutoNegLANPort**: this identifier can be set to YES or NO. YES means that the port uses auto negotiation to decide speed and duplex. If the parameter is omitted, the default value is YES.
- **SpeedLANPort:** this identifier can be set to 10 Mbit/s or 100 Mbit/s. This identifier is only used when **AutoNegLANPort** is set to NO.
- **DuplexLANPort**: this identifier can be set to FULL or HALF. FULL means that the LAN port can receive and send at the same time. This identifier is only used when **AutoNegLANPort** is set to NO
- **AutoNegPCPort**: this identifier can be set to YES or NO. YES means that the port uses auto negotiation to decide speed and duplex. If the parameter is omitted, the default value is YES.
- **SpeedPCPort:** this identifier can be set to 10 Mbit/s or 100 Mbit/s. This identifier is only used when **AutoNegPCPort** is set to NO
- **DuplexPCPort**: this identifier can be set to FULL or HALF. FULL means that the LAN port can receive and send at the same time. This identifier is only used when **AutoNegPCPort** is set to NO

**DuplexLANport** and **DuplexPCPort** should only be set to **FULL** if the remote LAN port to which the IP telephone is physically connected to is manually set to full duplex.

**Note:** If a server or switch port is set to full duplex, but the other end is trying to make auto negotiation, there will be a half / full duplex mismatch.

If the [AutoNeg] header is omitted, auto negotiation will be used.

If the GBit option unit is connected to the phone (only DBC43x and DBC444), these parameters are irrelevant. Auto negotiation is used and the LAN port and the PC port in the option unit are set according to the result of the auto negotiation. To obtain full duplex auto negotiation must be set in both the phone and in the LAN switch. If full duplex is set in one end the result will be half duplex.

## 3.8 AUTOMATIC CHECK FOR NEW FIRMWARE

The telephone will read the configuration file every 24th hour to check for new firmware. For not registered telephone this check is always done and if new firmware is available it will be loaded. For registered telephones this check can be enabled via the configuration file and new firmware will be loaded. If the telephone is busy, the check will be performed if the telephone becomes idle before the delay time elapses, otherwise the check will be done at next scheduled time.

The header is [AutoResync] and following parameters can be used:

- **Time= hh:mm**: The time when the check shall be done.
- **MaxDelay=**: Value range 0-1439 minutes. The maximum time that the phone waits past the scheduled time before starting the check. The actual check will be done after a random time between the times stated in the **Time** parameter and the **MaxDelay** parameter.

The default value is that the automatic check is disabled.

## 3.9 BACKUP GATEKEEPER

The backup gatekeeper is a gatekeeper that will be used if **Automatic Gatekeeper discovery** or the primary and secondary gatekeeper fails, depending on configuration. For more information see *Installation Instructions for DBC 43x and DBC 44x.*

The header for backup gatekeeper is [BackupGK]. The identifiers are:

- **System:** the type of backup gatekeeper. Possible values are BP-R16, MD-E, MD110 and MX-ONE. The backup gatekeeper **system** does not have to be the same as the main site **system**.
- **IPAddress:** the IP address of the backup gatekeeper.

## 3.10 CODEC PRIORITY

The supported codecs and their priority are defined below the header [Codecs]. The following dataless identifiers can be used:

- G.722
- G.711A
- G.711U
- G.729A
- G.729AB
- G.723

The identifiers must be written as a list with only one on each line and in priority order. The codecs that are not in the configuration file will not be used by the IP telephone. The default priority is as in the example above. G.711 µ-law is the American standard and G.711 A-law is the European.

G.722 is a wideband codec which can only be used in H.323 mode. This codec will only be used in non-gateway calls. It cannot be used with Mitel TSW (MD110) when fast start is used.

## 3.11 CORPORATE LOGON (ONLY DBC 444)

The header [CorporateLogon] is used to configure a list of servers/gatekeepers to logon remotely to.

The data identifiers are:

- **ServerAddress:** The IP address of the server.

- **ServerTxt:** The name of the server that will be displayed in a list.

  The text should be inside quotation marks. The following character sets are supported: Basic Latin (0x00-0x7F) and Latin-1 Supplement (0x80-0xFF). The maximum length of the string is around 21 characters.

The two data identifiers can be defined maximum 20 times.

**Note:** For more information about **Corporate Logon**, see Installation Instructions for DBC43x and DBC44x.

## 3.12 DEBUG LEVELS

The header [DebugLevel] is used to change the debug level for all the telephones via the configuration file for fault locating reasons. The following parameters exist:

**H323Debug**
The events that occur in the H.323 module in the telephone.

**UIDebug**
The events that occur in the user interface module in the telephone.

**RTPDebug**
The events that occur in the voice (media stream module) in the telephone.

**WAPDebug**
The events that occur in the WAP module in the telephone. This module handles services and display messages.

**WEBDebug**
The events that occur in the web server module in the telephone.

The following parameter values exist:

0 = No printouts

1 = Error printouts

2 = Major events

3 = Minor events

4 = All printouts

## 3.13 DEFAULT MELODY LIST

The header [DefaultMelody] is used to define a list of default melodies that can be used instead of ring signals. If the header is disabled no melodies are possible to define in the configuration file or in the menus in the telephone.

The following parameters are available:

- **MelodyListIndex**. 1 - 65535. This sequence number shall be increased each time the system administrator wants that the list with default melodies shall be updated in all the telephones.
- **Melody1**. **Melody10**. The position and file name of the melody that shall be included in the list of default melodies. The file extension is **.pcm**. The rules for the file name are:
  - max 25 characters including the extension.pcm
  - letters in the English alphabet, underscore, hyphen
  - digits 0-9
  - DELETE means that the melody will be erased in this position in all the phones
  - must not contain any space

Example: define a list with 5 default melodies and remove the melody on position 6 in all telephones. Position 7-10 can be used be the end-users individually.

```
[DefaultMelody]

MelodyListIndex=7
Melody1=IGotYou.pcm
Melody2=HereComesTheSun.pcm
Melody3=Rollercoaster.pcm
Melody4=Chehra.pcm
Melody5=RingRing.pcm
Melody6=DELETE
;Melody7=
;Melody8=
;Melody9=
;Melody10=
```

**Note**. If an end-user has defined a melody in a position that is defined in the configuration file, it will be overwritten when the configuration file is loaded into the phone and if the MelodyListIndex is increased compared to the previous time.

See also Installation Instructions for DBC 43x and DBC 44x.

## 3.14 DEFAULT PACKET SIZE

(Only H.323). It is possible to change the default packet size for the RTP stream. The used packet size is negotiated with the other end-point. When the telephone is master, the telephone will select the parameter value defined for the identifiers below, as the preferred packet size.

A shorter packet size means less delay and lower risk for echo, but means on the other hand more overhead and more traffic on the network.

The header is [DefaultPacketSize]. The identifiers are:

- **DefaultMsPerPacket711**: The size of the packets in ms for the G.711 µ-law and A-law codecs. Possible values are: 10, 20, 30 ms. The default value is 30 ms.
- **DefaultMsPerPacket729**: The size of the packets in ms for the G.729A and G.729AB codecs. Possible values are: 10, 20, 30, 40, 50 and 60 ms. The default value is 30 ms.

## 3.15 DIFFSERV SETTINGS

Diffserv is a re-interpretation of the ToS (Type of service) field used to give priority to the IP datagrams. The Diffserv octet consists of the field DSCP (Differentiated Services Codepoint, bits 0 to 5) and Reserved (bits 6 and 7). The reserved field is always 0. The DSCP field consists of the fields Traffic class (bits 0, 1, 2) and Drop precedence (bits 3, 4, 5). Bit 0 is the leftmost bit.

The Traffic class field may take the values 0 (Best effort), 1 (Class A), 2 (Class B), 3 (Class C), 4 (Class D), 5 (Expedited forwarding) and inherited from ToS, 6 (Internet control) or 7 (Network control). Traffic class 6 and 7 should never be used in the IP telephone.

The Drop precedence field may take the values 1 (Low drop precedence, bit value 2), 2 (Medium drop precedence, bit value 4) or 3 (High drop precedence, bit value 6). Low drop precedence means that the corresponding packet has a higher probability of surviving router congestions. When Traffic class is set to Expedited forwarding the Drop precedence field will have the value 110 independent of which value has been set in the configuration file.

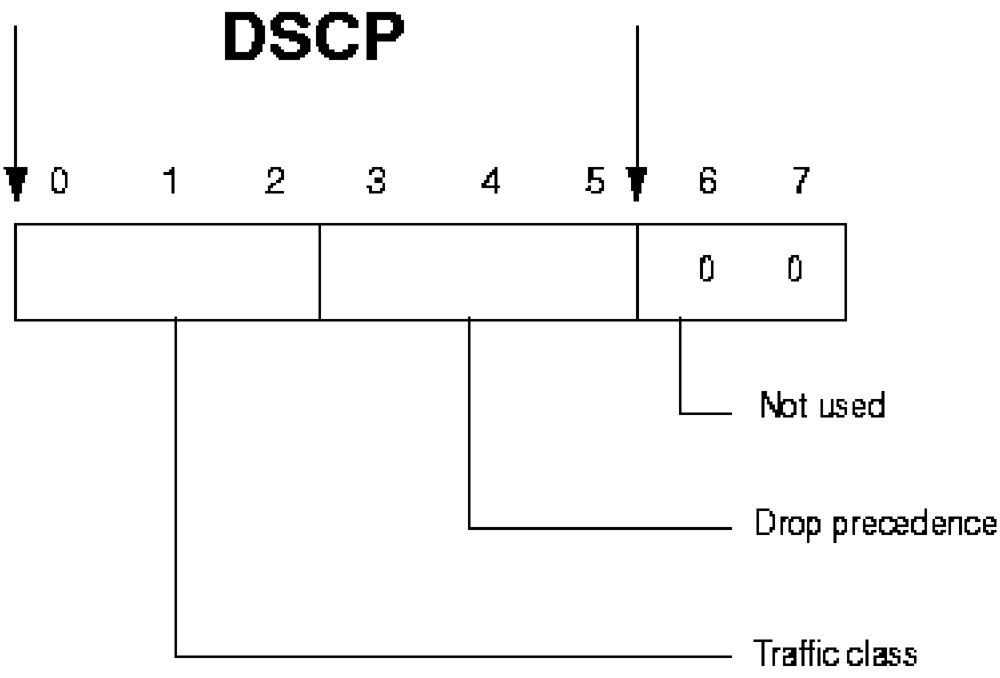



**Figure 2: The Diffserv octet**

The header [DIFFSERVUDP] is used to set the diffserv DSCP field in voice packets and in UDP signaling packets (RAS and WAP).

The header [DIFFSERVTCP] is used to set this field in TCP signaling packets.

These headers have two data identifiers used to set the Traffic class and the Drop precedence:

**Traffic Class**, this data identifier can be set to eight different values:

- 7 (Network Control bit 0-2: 111) Should not be used.
- 6 (Interned Control bit 0-2: 110) Should not be used.
- 5 (Expedited Forwarding bit 0-2: 101)
- 4 (Class D bit 0-2: 100)
- 3 (Class C bit 0-2: 011)
- 2 (Class B bit 0-2: 010)
- 1 (Class A bit 0-2: 001)
- 0 (Best Effort bit 0-2: 000)

**Drop Precedence**, this data identifier can be set to three different values:

- 3 (High Drop Precedence bit 3-5: 110)
- 2 (Medium Drop Precedence bit 3-5: 100)
- 1 (Low Drop Precedence bit 3-5: 010)

By default the setting is Expedited forwarding.

The default value for UDP is Expedited forwarding and the default value for TCP (H.225, H.245) packets is Class D and High drop precedence.

## 3.16 DISPLAY

The header [Display] is used to define how certain kinds of text or events shall be shown in the display. The data identifiers are:

- **ShowDTMFDigits:** defines if the entered DTMF digits shall be shown in the display or not. The following values are allowed:
  - **NO**. Digits will not be shown in the display.
  - **YES**. Digits will be shown and remain in the display.
  - **1-60**. Time in seconds that the entered digits will be shown in the display. Default value is 1 second.
- **BacklightMNSIncoming.** Indication of calls on an MNS key (not DBC 433).
  - **YES**. Independent of ring signal type for the MNS key, the display backlight is lit when there is a call on the MNS key. This is the default value.
  - **NO**. If type of ring signal is set to silence, the display backlight is **not** lit when there is a call on the MNS key.
- **BlinkOpuMNSIncoming.** When using an option unit OPU, it is possible to define when the extra bell or lamp shall be activated.
  - **YES**. The lamp or extra bell shall indicate incoming calls to the MNS keys. This is the default value.
  - **NO**. The lamp or extra bell shall not indicate incoming calls to the MNS keys.

- **MissedCallAtBusy.** When the user has disabled free on busy (he/she does not want to receive a second call while there is an ongoing call), it is possible to chose if the new call shall be visible as a missed call or not. The parameter values can be:
  - **YES**. Incoming calls while the terminal is busy shall be visible in the list with missed calls. This is the default value.
  - **NO**. Incoming calls while the terminal is busy shall **not** be visible in the list with missed calls.
- **MissedCallTimer = (0 - 1860 seconds).** Time in seconds before an incoming call in ringing state is regarded as a missed call.
  **0 =** the call is regarded as a missed call independant of the time in ringing state. This is the default value.

**For DBC 43X:**

The following parameter is handled in a special way in the DBC 43x telephones:

- **ShowIPSettings**
  - **ENABLED** The IP address for the user is shown in the display when pressing , selecting **Help** and **About Mitel 743Xip** -> **Phone IP Address**. Default value.
  - **DISABLED** The IP address is not shown to the user. The IP settings are only shown in the phone when logged on as an administrator.

**For DBC 444 and DBC 446:**

The following parameters are handled in a special way in the DBC 44x telephones:

- **ShowIPSettings**
  - **ENABLED** (for DBC 446) The IP settings (IP addresses, sub net mask, default gateway and so on) are shown in the display. Default value. The user can see these settings by pressing -> **Administrator Settings** -> **Network**.
  - **ENABLED** (for DBC 444)The IP address is shown in the display. Default value. The IP address for the telephone is shown when pressing , selecting **Help** -> **About** -> **Mitel 7444ip** -> **Phone IP Address**.
  - **DISABLED** The IP settings are not shown for the user. The IP settings are only shown if the administrator mode is entered in the phone.
- **AutoShowMonitorKeys:** defines if the page with monitor keys shall pop up. See also section 3.15 Shortcut keys.
  - **YES** The Monitor keys menu pops up when a call is received on the monitored extension or when the monitored extension makes an outgoing call.
  - **NO** The Monitor keys menu does not pop up. Default value.
  - **IncommingCall**. The Monitor keys page pops up only when the monitored extension gets an incoming call. The page is shown during the ringing phase and 5 seconds after answer.
  - **OutgoingCall.** The Monitor keys page pops up only when the monitored extension makes an outgoing call. The page is shown 15 seconds after off hook at the monitored extension.

## 3.17 EMERGENCY

(Only H.323). The header [Emergency] is used to set data to make it possible to initiate emergency calls from an unregistered telephone. It is also used to define emergency text to be shown for logged on telephones. The following identifiers are used:

- **System1 =** the system of the emergency call server or gatekeeper. This identifier supports the same systems as the **System** data identifier belonging to the header **[SYSTEM]**. See section 3.2 System.
- **Address1 =** the IP address of the emergency call server or gatekeeper.
- **Port1 =** the call signaling port number of the emergency call server or gatekeeper.
- **System2** = the system of the secondary emergency call server or gatekeeper.
- **Address2** = the IP address to the secondary emergency call server or gatekeeper.
- **Port2** = the call signaling port to the secondary emergency call server.
- **EmergencyNr =** the emergency telephone number, maximum 7 digits, e.g.: EmergencyNr=112. This number can be combined with the **RouteAccessNumber**, see section 3.29 WAP, to make it possible to dial for example 112 and 00112.
- **EmergencyNr2 =** (Only DBC 444)a second emergency telephone number can be defined.
- **EmergencyNr3 =** (Only DBC 444)a third emergency telephone number can be defined.
- **EmergencyTxt1 = "text"**. (Only DBC 444). This parameter can be used for two reasons:
  - Logged off telephone: the text row to be shown in the display associated with the **EmergencyNr**. If the default text shall be shown, this parameter shall not be defined, it shall only be defined if another text is wanted. The default text is **For SOS calls, dial xxx**.
  - Logged on telephone: the text row to be shown in the same text box as the follow-me information. See also the parameter **ShowEmergencyTxt** below

  The text must be inside quotation marks. The following character sets are supported: Basic Latin (0x00-0x7F) and Latin-1 Supplement (0x80-0xFF). The maximum length of the string is around 35 characters.
- **EmergencyTxt2 = "text".** (Only DBC 444). This parameter can be used for two reasons:
  - Logged off telephone: the text row to be shown in the display associated with the **EmergencyNr2**.
  - Logged on telephone: the text row to be shown in the same text box as the follow-me information. See also the parameter **ShowEmergencyTxt** below.

  The text must be inside quotation marks. The following character sets are supported: Basic Latin (0x00-0x7F) and Latin-1 Supplement (0x80-0xFF).The maximum length of the string is around 35 characters.
- **EmergencyTxt3 = "text".** (Only DBC 444). This parameter can be used for two reasons:

– Logged off telephone: the text row to be shown in the display associated with the **EmergencyNr3**.

– Logged on telephone: the text row to be shown in the same text box as the follow-me information. See also the parameter **ShowEmergencyTxt** below

The text must be inside quotation marks. The following character sets are supported: Basic Latin (0x00-0x7F) and Latin-1 Supplement (0x80-0xFF).The maximum length of the string is around 35 characters.

- **ShowEmergencyTxt =** (only DBC 444)
  - **YES**. The text rows defined by the EmergencyTxtn parameters will be shown in a text box in the idle menu when the telephone is logged on.
  - **NO**. The text rows defined by the EmergencyTxtn parameters will not be shown, when the telephone is logged on. This is the default value.
- **A-Number =** the telephone number sent to an emergency centre. The number can be used for dial back by the emergency centre. The number can be associated with a geographical area.
- **RouteId =** the identifier/password used by the emergency call server to identify the call from the telephone, maximum 15 ASCII characters (digits and letters).
- **NumberingPlanOfA-Number =** the type of numbering plan (according to Q.931) that is used for the A-number. Two values are allowed.

a) **ISDN**: This value is used when the emergency centre is in the public network. This is the default value.

b) **Private**: Private numbering plan is used for the A-number. This can be used for example when the emergency centre is inside the campus and the centre expects internal numbers.

For Mitel TSW an IP trunk board is used as emergency call server.

## 3.18 SHORTCUT KEYS

In this section the expressions TNS and MNS are used:

**TNS**: Telephony Name Selection. Phone numbers and function codes (e.g. *21#) can be assigned to TNS keys.

**MNS**: Phone number to a monitored extension.

The shortcut keys on DBC 434 and DBC 444 can be assigned as function keys (e.g. call back), as TNS keys and as MNS keys. The assignments of some types of the function keys are made from the configuration file, see below.

The assignments of the following types of shortcut keys are made from the PBX:

- MNS
- TNS but can also be done from a menu in the phone, or via the web interface in the phone.
- Malicious Call Trace (MCT)
- Recording key to start the recording of a call
- Personal Number (PEN) key

**Note:** When using MX-ONE Provisioning Manager (PM) it is possible to fetch and show the current programming of the shortcut keys. The first step is to set the terminal administrator password in PM:

**System** > **Subsystem** > create or change > **Terminal Password**

After this it is possible for PM to fetch and then show the key data.

### 3.18.1 DBC 434 AND DBC 444

The headers [FunctionKeysDBC434] and [FunctionKeysDBC444] in the configuration file are used for:

- Assigning functions (for example, call back or follow me) to the shortcut keys
- Defining if the TNS key data shall be stored in the phone or in the PBX.

The shortcut keys are numbered according to the figure 3 on page 23.

**Note:** When using DBC 434 and DBC 444 with MX-ONE SN and TSW and when the data associated to shortcut keys (TNS or MNS) is stored in the PBX, please consider the following:

To avoid future problems when adding or removing functions that are connected to shortcut keys, it is important to plan which function to associate with each shortcut key when performing new installation of IP phones.

If a function associated to a shortcut key is removed, all the existing TNS and MNS keys will be moved to the previous key number. This can be avoided by replacing functions that are to be removed with other functions.

If a function associated to a shortcut key shall be added, try to replace an existing shortcut key. If this is not possible all the existing TNS and MNS keys will be moved to the next key number.

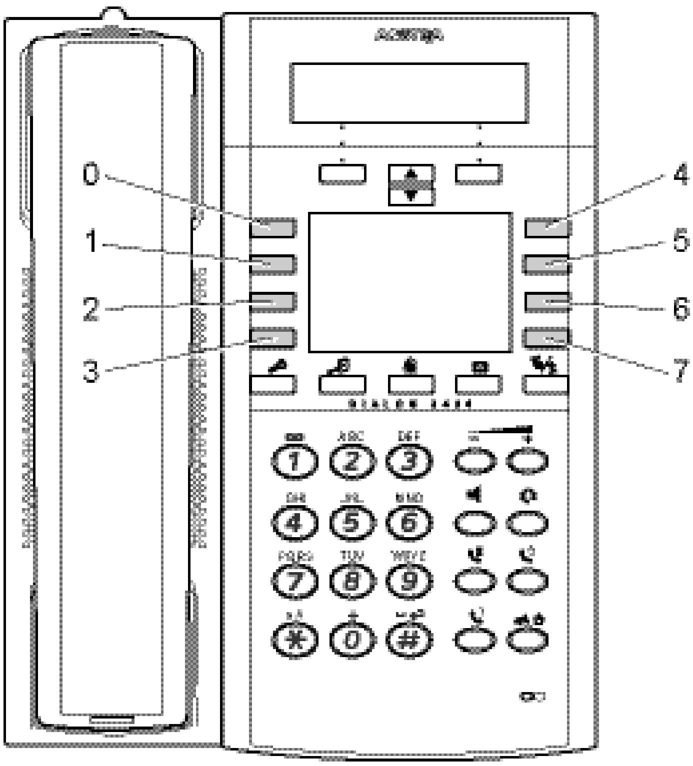



**Figure 3: Shortcut Keys on DBC 434 and DBC 444**

In DBC 444 there are 80 shortcut key positions available. There are up to nine virtual pages with shortcuts.

In DBC 444 there is a pop-up option for monitor keys, see section 3.13 Display. With this feature enabled, the page containing a monitor key is displayed when e.g. a call to the associated monitored extension is received.

It is possible to attach key panel units to DBC 434 and to DBC 444. Figure 4 below shows the key numbering on the key panel units.

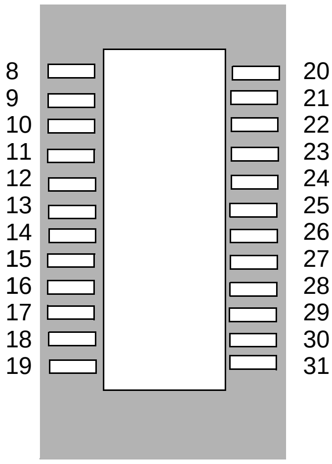



**Figure 4: Shortcut keys on DBY 412 11 (KPU) and DBY 412 21 (DPU)**

The table below shows which key number that corresponds to the key panels.

**Table 1 Key numbering on key panel units**

| Key panel units | Key number |
|---|---|
| 1 | 8-31 |
| 2 | 32-55 |
| 3 | 56-79 |

Maximum two display panel units (DPU) or maximum three key panel units (KPU) can be connected to the phone. KPU and DPU cannot be mixed on the same phone.

The headers [FunctionKeysDBC434] and [FunctionKeysDBC444] have the following data identifiers:

- **CallBack**
- **Transfer**
- **FollowMe**
- **FreeOnSecond**
- **Conference**
- **CallPickUp**
- **CallWaiting**
- **Intrusion**
- **DirectoryURL**

- **EnablePBXStoring:**
  - **YES**: store the TNS key data in the PBX. The software in the PBX must support the storing of TNS key data
  - **NO**: TNS key data is not stored in the PBX. Default value
- **RejectKey:**
  - **ENABLED**: The Reject softkey and hard key are available. This is the default value.
  - **DISABLED**: The Reject softkey will not be shown. The hard key (the Clear key while the phone is in ringing state) cannot be used for rejecting the call.

If the header is omitted the shortcut keys will be placed in their default position.

**Table 2 Shortcut keys, default values**

| Shortcut key | MX-ONE | BP | MD-E |
|---|---|---|---|
| Free on busy | 0 | - | - |
| Call back | 1 | - | - |
| Follow me | 2 | - | - |
| Corporate Directory | 3 *) | | |

*) Only in DBC444. If the corporate directory URL is enabled this key will by default show Corporate Directory.

If the header is enabled, every key assigned with a function must be defined in the configuration file.

If one key shall be removed in the default settings, all other keys must be defined. Example: the follow-me key shall be removed but the rest of the keys shall be in the same positions:

```
[FunctionKeysDBC434]
FreeOnSecond=0
CallBack=1
;FollowMe=2
DirectoryURL=3
;Conference=
;Transfer=
;CallPickUp=
;CallWaiting=
;Intrusion=
EnablePBXStoring=YES
```

## 3.18.2 DBC 446

There are eight shortcut key pages. See figure 5 on page 26 for the shortcut key numbering principle.

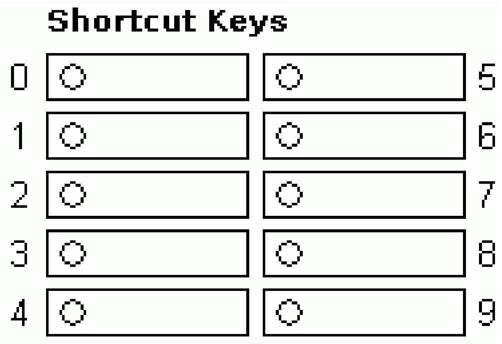



**Figure 5: Shortcut Key Numbering Principle**

There is a pop-up option for Monitor Keys, see 3.16 Display on page 19. With this feature enabled, the page containing a MNS key is displayed when e.g. a call to the associated monitored extension is received. When using this feature, it is recommended that all MNS keys are placed on the same page.

The header [FunctionKeysDBC446] is used to set if the TNS keys shall be stored in the PBX.

The data identifier is:

- **Enable PBX Storing**.
  - **YES**: store the TNS key data in the PBX
  - **NO**: TNS key data is not stored in the PBX. Default value

If **EnablePBXStoring** is set to **YES**, the software in the PBX must support the storing of TNS key data.

## 3.19 HEADSET RING TONE

If headset preset mode is used and if automatic answer is selected, it is possible to get a tone in the headset to announce that there is a new call. It is also possible to select if a ring signal shall be initiated or not.

The header [HeadsetPreset] has the following parameter:

- **HeadsetTone: YES** or **NO**. A tone in the headset announces a new call. The default value is **NO.**
- **SpeakerRinger: YES** or **NO.** A ring signal announces a new call. The default value is **YES.** In this case *auto answer with delay* must be used in the phone.

## 3.20 IP PHONE ADMINISTRATOR

The tool IP Phone Administrator is used to monitor registered and un-registered IP telephones.

For MX-ONE Service Node the tool is integrated in the MX-ONE Service Node Manager (SNM).

For Mitel TSW and other platforms the tool is a stand alone application.

The header [IPPhoneAdministrator] is used to define data for the IP Phone Administrator tool:

- **IPPhoneAdministror:** can be set to **YES** or **NO**. The value **YES** means that the telephones will sent http messages to the IP Phone Administrator server, if there is an IP address initiated to this server. The value **NO** means that the telephone does not send such messages. The default value is **YES**.
- **ServerAddress**. The IP address to the IP Phone Administrator server can be set manually in the configuration file. An alternative way to get this IP address is from a DNS SRV resource record. When using the telephone with MX-ONE SN, it is the IP address to MX-ONE Service Node Manager in LIM 1 that shall be defined.
- **ServerPort**. The port number to the IP Phone Administrator server can be set manually in the configuration file. An alternative way to get this port number is from a DNS SRV resource record.

For more information, see *Installation Instructions for DBC 43x* and *DBC 44x.*

## 3.21 LANGUAGE

The header [Language] has the following data identifiers:

- **LanguageFile**. The path and the file name of the language file for DBC 43x or DBC 446.
- **LanguageVersion**. The version of the language file for DBC 43x or DBC 446.
- **DBC444LanguageFile**. The path and the file name of the language file for DBC 444.
- **DBC444LanguageVersion**. The version of the language file for DBC 444.
- **OptionalLanguage**. Here can the procedure to change language in the gatekeeper also be stated, e.g. *08*3# (if the gatekeeper supports such a procedure). The languages stated here are shown in the language menu in the IP telephone.
- **StartupLanguage**. During the start up sequence only English is available, but when the telephone is running, another language can be used. This start up language is defined here.

  When the user has set the language in his telephone, using the menu or the procedure (e.g. *08*3#), this specific language will be saved in the telephone. After a restart the telephone will start up with this language, ignoring the start up language defined in the configuration file.

Translated languages:

AN - Lithuanian
CS - Czech
DA - Danish
DE - German
EN - English
ES - Spanish
FI - Finnish
FR - French
HU - Hungarian
IT - Italian
LT - Estonian
NL - Dutch
NO - Norwegian
PB - Brazilian Portuguese

PL - Polish
RO - Roumanian
RU - Russia
SK - Slovak
SL - Slovenian
SV - Swedish

## 3.22 LED CADENCE SETTINGS

Three different LED cadence settings can be set. The header [LEDs] has the following data identifiers:

- **Cad0:** in a call.
- **Cad1:** call parked.
- **Cad2:** incoming call.

For each of these three LED settings, four intervals must be set. First value is how long the LED should be on, the second is how long it should be off, the third is on and the fourth is off, then the first value is used again and so on. The values have the range 0-255, one unit is 10 ms. For default values see section 4 Examples.

## 3.23 LLDP-MED

The Link Layer Discovery Protocol (LLDP) is a vendor-neutral Link Layer protocol used by network devices for advertising their identity, capabilities, and neighbors. The Media Endpoint Discovery is an enhancement of LLDP, known as LLDP-MED. The DBC43x and DBC44x telephones can use this protocol to get the VLAN identity and priority.

This method to get VLAN identity overrides the settings in the configuration file and in DHCP option 43.

The header [LLDPMED] has the following data identifiers:

- **LLDPMED:** can have the two values **Enabled** and **Disabled.** The default value is enabled.
- **TTL:** Time between the telephone updates the LAN switch with LLDP information. Value 0-65535 seconds, where zero means not valid data. The default value is 120.

For more information see Installation Instructions for DBC43x and DBC44x.

## 3.24 LEVEL 2 QUALITY OF SERVICE

The header [L2QOS] is used in the IP telephone switch to:

- give different priorities for PC traffic and voice traffic (according to 802.1Q). See section 3.19.1 Virtual LAN settings.
- define VLAN tagging (according to 802.1Q) and set the VLAN identity. See section 3.19.1 Virtual LAN settings.
- limit the number of broadcast messages that the telephone shall handle. See section 3.19.1 Virtual LAN settings.

If the GBit option unit is connected to the phone (only DBC 43x and DBC444), all the parameters under this header are relevant.

### 3.24.1 VIRTUAL LAN SETTINGS

The switch has three ports:

- **LAN Port:** The port to the network.
- **PC Port:** The port to the PC.
- **Phone Port:** The port to the IP telephone.

The VLAN configuration file header is [L2QOS]. The following identifiers that can be set:

- **PhonePort=m,n:** this identifier has two values separated by a comma.

**m** User priority, see below. This value is used when the VLAN identity is defined in the configuration file or in DHCP option 43. The default value is 6.

**n** VLAN identifier, a number from 1 - 4094.

- **PCPort=m,n:** this identifier has two values.

**m** User priority, see below. The default value is 0. If the PC sends a priority value, the value from the configuration file will override the value from the PC, when the telephone forward the packet to the LAN.

**n** VLAN identifier, a number from 2 - 4094.
**Note:** The value 1 cannot be used.

- **LANPort=rs:** this identifier has two values.

**r** 0 = VLAN not used (untagged). 1 = VLAN used (tagged). When VLAN is not used the IP telephone always has higher priority than the PC.

**s** 0 = No VLAN for PC originating traffic (untagged).1 = VLAN will be used for PC originating traffic (tagged).

The user priority is defined as follows:

| | |
|---|---|
| **0** | Best effort, same as normal LAN traffic. |
| **1** | Background. |
| **2** | Spare, not recommended |
| **3** | Excellent effort. |
| **4** | Controlled load. |
| **5** | Video, less than 100 ms delay. |
| **6** | Voice, less than 10 ms delay. |
| **7** | Network control. |

For default values see the examples below.

Examples: If X is the voice VLAN and Y is the data VLAN. If only the voice traffic shall be tagged and the data traffic shall use the native LAN, use the following settings:

[L2QOS]

PhonePort=6,X

PCPort=0,Y

LANPort=1,0

If both the voice traffic and data traffic shall be tagged, use the following settings:

[L2QOS]

PhonePort=6,X

PCPort=0,Y

LANPort=1,1

It is not possible to have tagged traffic on the PC port but untagged traffic on the telephone port.

- **RememberVLAN**. This parameter defines the VLAN tagging of the telephone after a reboot.
  - **COLD:** After a power reboot (the power is disconnected) or software reboot, the telephone retains the previous VLAN identity. This option can be used in a network with VLAN in combination with IEEE802.1x or if a limited scope with IP addresses in the native LAN are available.
  - **WARM:** After a software reboot (the power is connected), the telephone retains the previous VLAN identity. After power reboot the telephone starts a new VLAN negotiation. This is the default value.

When the VLAN identity shall be changed, the RemeberVLAN parameter must not be set to COLD. If the parameter was previously set to COLD, the configuration file with the new parameter value must be downloaded to the telephones before the VLAN identity can be changed. When the parameter value is changed from COLD to WARM a new VLAN negotiation is started.

When defining the VLAN identity via the configuration file, it is important that the VLAN identities are correct, otherwise it is possible that the telephone accesses the native LAN and the VLAN in a loop.

The alternative to automatic VLAN detection is to edit the VLAN identity manually via the telephone menu or via the web interface

**Note:** If the VLAN identity shall be set manually, the header [L2QOS] with associated parameters have to be disabled in the configuration file.

### 3.24.2 BROADCAST MESSAGE LIMIT

With the broadcast message limit parameter it is possible to set the limit for number of broadcast messages that the switch in the telephone shall handle. This can be useful if broadcast storms occur on the LAN.

- If the limit is too high, there is a risk that the telephone freezes when trying to handle all the messages. If the limit is too low there is a risk that messages important for the telephone traffic are lost, which can mean that calls can be lost.
- **BroadcastStormControlLimit:** Define the number of broadcast messages to handle within 100 millisecond. The default value is 80 which mean that 800 broadcast messages per second are handled by the telephone.

## 3.25 MESSAGE

The header is [Messages]. (Only DBC 43x and DBC 444).

The Messages menu is accessed by pressing the key. There are two options: **My VoiceMail** and **My Messages**. The menu can be changed with the following parameters:

- **DirectVoicemail:**
  - **YES:** the voice mail is reached directly.
  - **NO**: the options **My VoiceMail** and **My Messages** are shown. This is the default value.
  - **VoiceMailNumber:**

    Defines a default Voice Mail Box Number

    E.g.  VoiceMailNumber=203

    In this example, when there is no message from MX-ONE or manual message waiting, pressing will call 203 directly without showing messages menu.

## 3.26 OMD SETTINGS (MX-ONE SERVICE NODE ONLY)

The header [OMD] is used to set the signaling address to the telephony server when DBC 43x or DBC 44x is used as an OMD (Operator Media Device). Two identifiers are used:

- **SNAddress =** the IP address of the telephony server.
- **SNPort =** the receiving OMD signaling port number of  the telephony server.

The identifiers for the address and the port number can be defined twice,  if the redundancy feature is used.

Example:

[OMD]

SNAddress=130.100.188.111

SNPort=1700

SNAddress=130.100.86.18

SNPort=1700

It is only the following configurations in the configuration file that are  relevant when the terminal is used as an OMD:

- Software
- Codec priority
- Tone configuration
- LED cadence settings
- Tone ringer cadence settings
- VLAN settings
- Auto negotiation
- 802.1x
- IPPhoneAdministrator

- Time
- WebServer.

The OMD can only be used for MX-ONE Service Node. The OMD is initiated in the telephony server with the *OPSAI* command, see the command description for *PBX OPERATOR TRAFFIC*.

## 3.27 PASSWORDS

The header [Password] is used to define how passwords and Personal Identity Numbers (PINs) are handled in the phone.

**Note:** For information on how to set the administrator password, see section 3.2 System.

Available data identifiers:

- **ServiceCodeChangePIN:** The service code for changing the PIN. Example *74*
- **PINorPassword:** It is possible to choose if a password or PIN shall be used when registering the phone to the system.
  - **PIN:** Shall be selected when the phone is used with MX-ONE 3.2 or later. The menu for changing the PIN can be selected. It is only possible to enter digits as PIN. The text string when registering the phone is *Enter PIN* instead of *Enter password*.
  - **Password:** The menu for changing PIN is not available. The text string at log on is *Enter password*. Default value.
- **PasswordInputFormat:** This parameter has only effect when PINorPassword=Password
  - **Alphanumeric:** When entering the password at registration of the phone, alphanumeric characters are allowed. Default value.
  - **Digit:** When entering the password at registration of the phone, only digits are allowed.
- **UseAdminPassword**: (DBC 433, DBC 434 and DBC 444 only). This parameter is used to set if the administrator password must be entered after pressing the keys ,*, and *5* simultaneously for at least one second to enter the administrator mode in the telephone.
  - **Enabled**: Administrator passwords has to be entered to get into administrator mode
  - **Disabled**: No administrator password is needed. This is the default setting.

## 3.28 POWER SAVING

The telephones have the following options for power saving:

- Switch off the terminals at certain times via the configuration file. Only DBC 434 and DBC444. The user has to switch on the telephone before using it again.
- Switch off the backlight at a certain time. Only DBC 444 and DBC44601. These are the only phones with slight backlight on, all the time.

The header [PowerSaving] has the following parameters:

- **PowerDownTime**=hh:mm. The time when the telephones shall be switched off. If a telephone is in speech mode when it shall be switched off, it will remain on until next the switch down time
- **BrightnessOffTime**=hh:mm. The time when the backlight shall be totally switched off.
- **BrightnessOnTime**=hh:mm. The time when the backlight shall be switched on.

The default values are that the terminals are on and the backlight is also on all the time.

## 3.29 VOICE RECORDING

It is possible to record calls to a central recording equipment. There are two options:

- Active recording: all calls to the monitored extensions are recorded.
- Record on demand, ROD: the user can start and stop the recording by pressing a function key.

The header [Recorder] has the following data identifiers:

- **LoggerIPAddress1**. The telephone checks that the IP address from which the recording is ordered corresponds to the IP address in this parameter value. If it does not correspond the recording will not start. The reason for this check is to improve the security to make it more difficult with illegitimate recording. Three IP addresses can be defined.
- **LoggerIPAddress2**. See LoggerIPAddress 1 above.
- **LoggerIPAddress3**. See LoggerIPAddress 1 above
- **LoggerIPAddress4**. See LoggerIPAddress 1 above
- **LoggerIPAddress5**. See LoggerIPAddress 1 above
- **LoggerIPAddress6**. See LoggerIPAddress 1 above

Recording on demand settings when using a NICE® recording system:

- **RODServerIPAddress**. The IP address to the server to which the telephone sends the record on demand requests.
- **RODServerName.** The name of the ROD server. This data must be provided by the administrator for the NICE Perform® recording system.
- **RODServerCredentials=username:password**. The login credentials towards the ROD server. The format of the parameter value must be as described above. This data must be provided by the administrator for the NICE Perform® recording system.
- **RODServerGKId**. The telephone must send this identity to the ROD server. This data must be provided by the administrator for the NICE Perform® recording system.

Recording on demand settings when using an ASC recording system:

- **RODStart.** URL sent by the phone to the ASC recording system when the recording key is pressed to start the recording. Example: http://192.105.88.152:8080/XVOIPService?page=START&OPN=n

  where n is the extension number added by the telephone. The IP address and port number must match the recorder's listening IP address and port number.

- **RODStop**. URL sent by the phone to the ASC recording system when the recording key is pressed to stop the recording. Example: http://192.105.88.152:8080/XVOIPService?page=STOP&OPN=n where n is the extension number added by the telephone.

Recording on demand settings when using another vendor of recording system:

- **RODStart.** URL sent by the phone to the recording system when the recording key is pressed to start the recording. Example: http://192.105.88.152:80/recbutton?command=start&user=n where n is the extension number added by the telephone. The phone needs the IP address but the rest of the URL can be specified according to what the recording system requires.
- **RODStop**. URL sent by the phone to the recording system when the recording key is pressed to stop the recording. Example: http://192.105.88.152:80/recbutton?command=stop&user=n where n is the extension number added by the telephone. The phone needs the IP address but the rest of the URL can be specified according to what the recording system requires.
- **RecordingTone**

  -ENABLED. A defined tone will be heard when call is recorded.

  -DISABLED. Default value.

For more information see Installation Instructions for DBC 43x and DBC 44x.

## 3.30 RINGLEVEL

- **MNSBusyRingLevel.** Used to control incoming MNS ring level at speech state.

- **Normal.** Default value. Latest user programmed ring level will be used.

**- LOW.** Minimum ring level will be used.

**- SILENCE.** No ring signal will be heard.

## 3.31 RING CADENCES

The header [RingCads] has the following data identifiers:

- **Internal=** ring signal when receiving internal calls.
- **External=** ring signal when receiving external calls.
- **Callback=** ring signal when the callback function is used.
- **Extra=** ring signal for other purposes.

For each of these settings, six intervals must be set. The intervals have the same meaning as for the LED cadence settings, but with two additional intervals. The values have the range 0-255, one unit is 50 ms. For default values see section 4 Examples.

- **Delay=** the alerting delay before the ring signal starts after the call is received. The delay is valid for alert 1st call, alert 2nd and 3rd call and for MNS keys. It is not valid for call back.The delay time is in second. The default value is 7 seconds.
- **CallWaiting=** When the telephone receives call waiting, the alerting can be of two types with the following parameter values:
  - **Tone**. Call waiting tone

- **RingSignal.** Ring Signal specified in *alert 2nd and 3rd call* in the phone. This is the default value

# 3.32 SECURITY

## 3.32.1 SECURITY IN H.323 MODE

The header [Security] is used to define the parameters for security in the telephone. The telephone has support for signaling encryption with TLS and support for media encryption with SRTP (Secure RTP). It is possible to load other certificates than those that are permanently stored in the telephone.

If the security is enabled and a valid password provided, the telephone will at registration try to use TLS for encryption of the signaling. The RAS signaling will use TCP instead of UDP. If the TLS/TCP negotiation fails, it is the security policy in the system and the other security parameters below, that decides what will happen. Signaling encryption according to TLS is also used in the call set up phase (H.245 and H.225).

If security is enabled, the telephone will try to use SRTP for media encryption.

- **Security=**
  - **ENABLED** If a valid password is provided, the telephone will try to use signaling encryption with TLS and if this is successful it will announce that it has support for media encryption with SRTP.
  - **DISABLED** The telephone will not announce TLS and SRTP capability. (Default value)
- **SecurityFallback=**
  - **YES**: If the TLS/TCP negotiation fails, it shall be permitted to continue the registration with RAS over UDP in a not secure way. (Default value).
  - **NO**: It shall not be permitted to continue the registration if the TLS/TCP negotiation fails.
- **CertificateValidate=**
  - **YES**: The telephone shall validate the server certificate. If the parameter value is YES, but the server does not have a valid certificate that is signed by one of the Certificate Authorities (CA) supported by the telephone, this will result in failed authentication. (Default value).

    **Note**. The parameter NTP= must be set, see section 3.27 Time. The reason is that the terminal must have correct time before it is registered.
  - **NO**: The telephone shall not validate the server certificate.
- **SaveUserPassword=**
  - **YES**: The end-user password (used to register towards the gatekeeper) is stored in the memory in the telephone in the same way as when security is disabled. (Default value).
  - **NO**: The end-user has to enter the password each time the telephone tries to register towards the gatekeeper, after power failure, network problems, firmware upgrade etc.
- **SignalingEncrypted=**
  - **YES**: The signaling is encrypted. This is the default value.

- **NO**: Null cipher which means that no encryption of the signaling is done but the signaling sequence is the same as when the signaling is encrypted. This option could be used at fault locating.

- **RootCert=**
  - **file name.pem**. The file name of the root CA certificate stored on the software server. The path where the file shall be stored must be according to the Installation Instructions for DBC43x and DBC 44x. The parameter is only used when a new certificate which does not already exists in the application shall be loaded.

    It is possible to remove a certificate that has earlier been loaded, by comment the file name. Example:
    ;RootCert=cacert.pem
- **UDPFilter=**
  - **ENABLED** All UDP ports that are not used in the call are blocked. Default value.
  - **DISABLED** All UDP ports are open.
- **UDPRateLimit=**

  With this parameter it is possible to set the rate limit for incoming UDP packets. If the limit is exceeded within 2 seconds, the additional packets will be discarded. Then new counting will start at the start of the next two seconds.

  Too low value can cause choppy speech, especially when short packet size (10 or 20 ms) is used.

  The default value in the application is 250 packets per two seconds.
- **TCPFilter=**
  - **ENABLED** All TCP ports that are not used are blocked. Default value.
- **DISABLED** All TCP ports are open.
- **OpenTCPPort1**
  - 1720. Default open port.
- **OpenUDPPort1**
  - 9200. Default open port
- **OpenUDPPort2**
  - 9200. Default open port
- **OpenUDPPort3**
  - 9200. Default open port

## 3.33 SERVICE CODES

The header [ServiceCodes] is used to define service codes sent from the telephone to the PBX. This is used for menu support of a procedure which means that the end-user does not have to remember the service code.

- **CancelCallBack** (DBC 43x and DBC 444 only)

This parameter is used to set the procedure for cancelling the pending call back missions. The parameter value can contain *, # or digits. Example: CancelCallBack=#37#

- **CancelManualMessageWaiting** (DBC 43x and DBC 444 only)

  This parameter is used to set the procedure for cancelling the pending message waiting missions. The parameter value can contain *, # or digits. Example: CancelMessageWaiting=#31#

## 3.34 SNMP AGENT

The header [SNMP] is used to manage the built in SNMP agent in the telephone. The following parameters exist:

- **SNMPAgent**. Used to enable/disable the SNMP agent in the telephone. The parameter values can be:
  - **DISABLED**. The default value.
  - **ENABLED**.
- **CommunityString**. When the agent is enabled, the system administrator must set this text string. The community string is used as a password.

  If the community string in the phone is equal to the community string set in the scanning tool, the telephone responds with the requested information. If the community string is incorrect, the telephone simply discards the request and does not respond.

For more detailed information about the SNMP agent, see installation instructions for DBC 420.

## 3.35 STORING USER SPECIFIC DATA ON A SERVER

The header [StoreUserData] is used to store the phone's user specific data on a SFTP or a FTP server. The following data can be stored on the FTP server:

**For DBC 43X, DBC 444 and DBC 446 from the R3A application**

- Contacts
- Call List
- Shortcut keys
- Own phone numbers (mobile, voice mail).
- Phone settings such as alerting type, ring level, selected language, free on busy, missed call status.
- MNS keys disabled for dial out (DBC444 only) e.g. using command setMNSKeyNoDial (see SSH command Help).

This file has the extension type .xml.

**For DBC 446 before the R3A application and DBC 42x telephones:**

- Contacts

This file has the extension type .txt.

**The following data identifiers are used:**

- **EnableStoring:** can be set to **YES** or **NO**. The default value is **NO**.
- **CompatibleWithD4:** can be set to **YES** or **NO**. Default value is **YES**. If YES, two files are saved on the FTP server; the contacts file (.txt) and the user data file (.xml). This option is used for compatibility reasons when DBC 43x and DBC 44x telephones are mixed with DBC 42x telephones.

  The xml file option is introduced in DBC446 from application R3A, which means that in a site with an DBC446 application earlier than R3A, the parameter value shall be set to **YES.**

  If the parameter is set to **NO**, it is only the user data file (.xml) that is saved on the FTP server. There are only DBC 43x and DBC 44x telephones at the site, no DBC 42x telephones.
- **IPAddress:** The IP address to the FTP server. The default value is the IP address to the software server where the FTP server also can be located.
- **FTPUname:** The user identity (or user account) under which the Contacts files are stored on the FTP server. Maximum 24 characters, the default user name is *Telephone*.
- **FTPPword:** The password for the user name (or account) under which the Contact files are stored on the FTP server. Maximum 24 characters, the default password is *Telephone*.
- **UseSFTP:** The type of server to be used when storing the phone book.
  - **YES**: Use a SFTP server for storing the files for user specific data.
  - **NO**: Use a FTP server for storing these files. This is the default value.

## 3.36 SUFFIX SERVICES

### 3.36.1 SUFFIX SERVICES

In H.323 mode this feature is only for MX-ONE. The header [SuffixServices] is used to setup services that can be used by pressing one digit after a call has been made to a busy extension, except for callback which can be initiated even if the extension is not busy. The following data identifiers can be used:

- **Intrusion** (value for the Standard application system = 4)
- **CallWaiting** (value for the Standard application system = 5)
- **CallBack** (value for the Standard application system = 6)
- **CallPickUp** (value for the Standard application system = 8)

Permitted values are: digits 0-9.

The identifiers shall be set to the digit which shall be used to initiate the service: e.g. CallBack=5.

## 3.37 TIME

The time that is shown in the telephone display can be updated via:

- WAP messages. The header [Time] has no effect.

- An SNTP server. SNTP (Simple Network Time Protocol) is a standard protocol used to retrieve the time in a LAN network.

The header [Time] shall only be used if time is updated via SNTP. If the parameter is omitted, the phone uses the time received in the WAP signals from the gatekeeper (only H.323). The header has the following data identifiers:

- **TIMEZONE** =id:D1:D2:D3:D4.
  - id= name of the time zone created. Examples:
  - CET = Central European Time
  - D1 = time in minutes east of UTC (Universal Coordinated Time) which also known as Greenwich mean time (GMT).
  - D2 = time in minutes west of GMT time
  - D3 = daylight savings time begins (month-day-hour)
  - D4 = daylight savings time ends (month-day-hour)

  If no daylight savings time shall be used, D3 and D4 shall be omitted.
- **NTP-server**. The IP address to the SNTP server from where the IP telephone shall fetch the time. When the system is MX-ONE and when the telephone does not get any response from the NTP server defined in this parameter, the phone will try to get the time from the primary and secondary gatekeeper.

If different time zones shall be used in a system, the time shall be received through SNTP.

When the telephone is used with MX-ONE SN in H.323 mode, time is fetched from:

- NTP server
- Primary gatekeeper
- Secondary gatekeeper

If security is enabled and certificate validated, time must be fetched from a NTP server.

## 3.38 TONE CONFIGURATION

The header [Tones] has the following data identifiers:

- Dial tone
- Special dial tone
- Busy tone
- Alerting tone
- Congestion tone
- Special information tone (number unobtainable)
- Call waiting tone
- Offhook queuing tone
- On hold tone
- External dial tone
- Recording Tone

- Waiting voice tone (also called *Connection in progress tone*). This tone can be used in traffic cases when it takes some time to establish the speech channel, for example answering on a monitoring key. The tone starts when the key is pressed and stops when the speech channel is established.
- Headset ring tone. See 3.19 Headset ring tone on page 26

Each tone has 16 values that must be set. The values have the following meaning:

1) Number of cadences, 0 means constant tone.
2) Number of frequencies, 0 means one frequency, 1 means 2 frequencies modulated, 2 and 3 are not implemented and 4 means three frequencies with one cadence each.
3) First tone on, time in milliseconds.
4) First tone off, time in milliseconds.
5) Second tone on, time in milliseconds.
6) Second tone off, time in milliseconds
7) Third tone on, time in milliseconds.
8) Third tone off, time in milliseconds
9) First frequency in Hertz (minimum 300, maximum 2000 Hertz).
10) Tone level for the first frequency, cannot be changed, it is always -10 dBm0.
11) Second frequency in Hertz (minimum 300, maximum 2000 Hertz).
12) Tone level for the second frequency, cannot be changed, it is always -10 dBm0.
13) Third frequency in Hertz (minimum 300, maximum 2000 Hertz).
14) Tone level for the third frequency, cannot be changed, it is always -10 dBm0.
15) Tone in voice, 0 is off, 1 is on. A tone locally generated in the telephone is mixed with the received RTP stream. Tone in voice = off, means that if a tone is played in voice, the voice will be switched off until the tone stops.
16) Tone not sine wave, only 0 is implemented which means sine wave.

To get a silent dial tone, value number nine (frequency of the first interval) should be set to zero. For default values see section 4 Examples. Tone level cannot be changed, it is always -10 dBm0.

**Note:** Only parameter values with integers can be used (no decimal point is allowed).

## 3.39 WAP

The header [WAP] has the following data identifiers:

- **MaxAttempts**. Maximal number of attempts to resend a WAP message if no answer is received. WAP messages are sent on unreliable UDP which means that re-sending is necessary at loss of packages. Minimum is 1 and maximum is 4.
- **RetransPeriod**. The time in milliseconds between each re-sending. Minimum is 2000 and maximum is 7000 with 1000 ms steps.
- **CountryCode**. The country code of the country where the PBX is located.
- **RouteAccessNumber**. The route access number.

- **CountryCodePrefix**. The prefix of the code of the country where the PBX is located.
- **NumberWithAreaCode:** if the area code shall be sent or not. This parameter is only used for calls within the own country.
  - **YES** include the area code and area code prefix when converting international phone number to dialling number. This is the default value.

    Example: +46 8 56867000 converted to 00 08 56867000, where 00 is the route access number.
  - **NO** remove the area code and area code prefix if the number is within the same area as the PBX, that is the area code in the number to dial is equal to the AreaCode in the configuration file. If the area code does not match the AreaCode in the configuration file, the area code will be sent.

    Example: +46 8 56867000 is converted to 00 56867000 when the AreaCode=8 in the configuration file.
- **AreaCodePrefix**. The prefix of the code of the area in the country where the PBX is located.
- **AreaCode**. The remaining part of the area code when the area code prefix is removed.

  Example: If the total area code is 08, the AreaCodePrefix=0 and the AreaCode=8.

**Match a dialed number to get the name from Contacts**

When a dialed number shall be matched with the number in Contacts to display the corresponding name, the following is valid:

If the area code is optional when making calls, the parameter AreaCode must be given.

Example: In Contacts the name and number + 46 8 568 6700 Peter Smith is stored (where 08 is the complete area code). The end-user dials either 00 08 568 6700 or 00 568 6700 (where 00 is the route access code) because the area code is optional. If the telephone shall be able to match this dialed number with the number stored in Contacts, to display Peter Smith, the AreaCodePrefix=0 and AreaCode=8 must be defined in the configuration file.

**Compose a B-number**

The CountryCode and RouteAccessNumber are used to compose a B party number to dial out via the PBX when a telephone number received from the corporate directory WAP server is a global international telephone number that has the format plus sign (+) CountryCode national telephone number.

If the country code in the received number is equal to the CountryCode in the configuration file, the plus sign and the country code will be replaced by the RouteAccessNumber. For the area code there are the following options:

- If NumberWithAreaCode=YES, the AreaCodePrefix and AreaCode will be added. Example: +46 8 56867000 will be dialed as 00 08 56867000.
- If NumberWithAreaCode=NO, the AreaCodePrefix and AreaCode will not be added. Example: +46 8 56867000 will be dialed as 00 56867000 if the AreaCode=8 in the configuration file.

If the country code in the received number is not equal to the CountryCode in the configuration file, it will not be removed, but the plus sign will be replaced by the RouteAccessNumber and CountryCodePrefix. The B party number to dial will consist of RouteAccessNumber, CountryCodePrefix, country code and the national telephone number. Example: +46 8 56867000 will be dialed as 00 00 46 8 56867000 if the country code is not equal to 46 in the configuration file. In this case the parameter NumberWithAreaCode does not have any impact.

## 3.40 WEB BROWSER

The header [WEBBrowser] is used to define parameters for the corporate directory in all phone types and for the web browser in the DBC 446 phone. The corporate directory uses XML with two options:

- XML interface for DBC 43x and DBC44x in CMG.
- a sub set of the XML interface for Mitel SIP Phones.

The header [WEBBrowser] has the following data identifiers:

- **HomeAdress**. (Only DBC 446). The URL to the home web page (portal), for example *http://www.mitel.com*.
- **DirectoryAddress**. (Only DBC 43x and DBC 446). The URL of the corporate directory server, see table 3 URL for different corporate directory servers on page 43
- **DBC444DirectoryAddress**. (Only DBC 444). The URL of the corporate directory web server, see table 3 URL for different corporate directory servers on page 43.
- **WebProxyIP**. (Only DBC 446). The IP address to the proxy server, for example *192.168.0.1*.
- **WebHttpPort**. (Only DBC 446). The http port number used by the proxy server, for example *8080*.
- **WebHttpsPort**. (Only DBC 446). The https port number used by the proxy server.
- **WebDNS1**. (Only DBC 446). IP address of the first DNS server, for example *192.168.0.1*.
- **WebDNS2**. (Only DBC 446). If the first DNS server does not work, use the second one.
- **NoProxyAddress**. (Only DBC 446). Bypass list. The address of the web sites within the intranet. The address are separated by commas, for example *internal.mitel.com,192.168.0.1*.
- **UserAgent**. (Only DBC43x and DBC 446). User Agent of the browser. The Corporate Directory server uses it to identify the correct telephone type. Parameter values:
  - **DBC 43x01** or **DBC 44601** (default values). Shall be used towards CMG.
  - **Aastra55i<MAC><VERSION>**. Can be used for other directory servers than CMG. This string is sent in the http GET message header. The phone will replace <MAC> with the MAC address of the phone and <VERSION> with the version of the XML API, see installation instructions for DBC 43x and DBC 44x.
- **DBC444UserAgent**. (Only DBC444). User Agent of the browser. The Corporate Directory server uses it to identify the correct telephone type. Parameter values:
  - **DBC 44401** (default value). Shall be used towards CMG.

   - **Aastra55i<MAC><VERSION>**. Can be used for other directory servers than CMG. This string is sent in the http GET message header. The phone will replace <MAC> with the MAC address of the phone and <VERSION> with the version of the XML API, see installation instructions for DBC 43x and DBC 44x.
- **UseWEBBforCorpDir**: **YES** or **NO** (default value). (Only DBC 446). Use the web browser to access the corporate directory. The value YES shall only be used for backward compatibility reason but not for new installation.

For more information on web browser settings, see *Installation Instruction for DBC 43x and DBC 44x*.

**Table 3 URL for different corporate directory servers**

| Corporate Directory | URL in DirectoryAddress and in D444DirectoryAddress |
|---|---|
| CMG *) | <IP address:port>/CorpDir/d4/d4.aspx<br>or<br><DNS name:port>/CorpDir/d4/d4.aspx |
| CMG **) | <IP address:port>/CorpDir/default.aspx<br>or<br><DNS name:port>/CorpDir/default.aspx |
| XML for Mitel SIP phones | dependent on the directory server |

*) using the XML interface in DBC 43x and DBC 44x.

**) using the web browser in DBC446

## 3.41 WEB INTERFACE

The header [WebServer] is used to enable or disable the web interface for the end-user.

From a web-browser in a PC, it is possible to log on to the telephone for handling of the contact list, call list, and so on. It is possible to log on as an end-user or as a system administrator. One parameter exists:

- **webInterFaceForUser**:
   - **Enabled**: the end-user web interface is enabled. This is the default value.
   - **Disabled**: the end-user web interface is disabled. The web interface for the system administrator is not affected.
   - **Enabled_with_default_pwd**: if the end-user has a PIN or password, they are used for logging in to the web interface. If the end-user has no PIN or password it is possible to log in with the default password *Welcome*.

**Note:** When using the option **Enabled_with_default_pwd**, and the user does not have any PIN or password, it is very easy to log in to someone else's telephone and look in the call list and contacts.

# 4 EXAMPLES

## 4.1 DBC 444 AND DBC 446 USED WITH MX-ONE

In this example DBC 444 and DBC 446 telephones are used with MX-ONE SN.

```
[System]
;--------------------------------------------------------------
; File:  d44x01-config.txt
; Product id and version: CAA 158 0058 R3F
; Release date: 2011-06-29
;--------------------------------------------------------------
; The system parameter may have the values MD110, MX-ONE, MD-E,
BP-R15 or BP-R16
;System=BP-R16
;System=MD110
;System=MD-E
System=MX-ONE
OverLap=YES
RRQTtl=600;only for H323
GatekeeperID=LIM*
;Domain=aastra.se
;PrimaryGKAddress=192.168.0.1
;SecondaryGKAddress=192.168.0.1
GatekeeperDiscovery=NO
;AdminPasswd=RQzceyb9c9
;LogOffRestriction=LogOffAllowed
;LogOffRestriction=DefaultNumberUsed
;LogOffRestriction=PermitIndividualLogOn
;LogOffTime=03:00
;OperatorNumber=09
```

```
[Software]
;----------------------------------------------------------
; The following files are affected:
; Application       d44x01_Applic_Rev.dat    (CAA158 0057)
; Application       d44401_Applic_Rev.dat    (CAA158 0070)
; Config File       d44x01-config.txt        (CAA158 0058)
; Language File     d44601-lang_Rev.txt      (CAA158 0059)
; Language File     d44401-lang_Rev.txt      (CAA158 0072)
;----------------------------------------------------------
```

```
ApplicationFile=/dbc44x01/d44601-applic_R2H.dat
ApplicationVersion=R2H
DBC444ApplicationFile=/dbc44x01/d44401-applic_P1A3t2.dat
DBC444ApplicationVersion=P1A3t2
```

```
;[802.1x]
;LANAccessControl=AUTO
;StoreUserIdPassword=YES
;UserType=PHONE;PHONE or USER
;UserIdentity=
;UserPassword=
;EAP_TLS=ENABLED
;RootCert=cacert.pem
;ClientCert=clientcert.pem
```

```
;ClientKey=clientkey.pem
;ClentKeyPassw=aastra

[Absence]
Profile=*10*profileNo#,#10#
FollowMe=*21*extensionNo#,#21#
ExtFollowMe=*22#extensionNo#,#22#
AbsenceReason=*23*reasonCode#,#23#
Lunch=0*hhmm
Busy=1
Absent=2*hhmm
Meeting=3*hhmm
Trip=4*mmdd
Course=5*mmdd
Vacation=6*mmdd
DayOff=7*mmdd
GoneHome=8*hhmm
Illness=9*mmdd
;OfficialMatter=2*hhmm
;FreeTime=6*mmdd
;Special=9*mmdd
;ParentalLeave=

[Authorization]  ;replaces the following code with "-"
Code=*61*        ;Account code for Standard Appl. System
Code=*72*        ;Auth. code for Standard Appl. System
Code=*73*        ;Lock Central auth. code
Code=#73*        ;Unlock cent. auth. code
Code=*74*        ;Change individ. ath. code
Code=*75*-*n#    ;Dial with indiv. auth. code. n means that
                 the nr will be visible
Code=*76*        ;Lock individ. auth. code
Code=#76*        ;Unlock individ. auth. code

[AutoResync]
Time=03:00
Delay=60

[AutoNeg]
AutoNegLANPort=YES;When AutoNeg is set to YES, Speed and
;DuplexLANPort=HALF;Duplex will not be read. The resulting
;SpeedLANPort=10;speed and duplex will be the result of
AutoNegPCPort=YES;the negotiation between the phone and the
;DuplexPCPort=FULL;network device it is connected to
;SpeedPCPort=100;/switch/hub/PC).
;network device it is connected to
;/switch/hub).
;[BackupGK]
;System=MD-E
;IPAddress=192.168.0.1

[Codecs]
;G.722  ;OBS: TSW (with fastConnect) doesn’t allow calls to be
;made if G.722 is configured. Works only for IP to IP calls.
G.711A; G.711 is mandatory but the priority order is optional
G.711U
G.729A
G.729AB
G.723
```

```
;[CorporateLogon]
;only for H323.
;ServerAddress=192.168.0.1
;ServerTxt="Server1 name"
;ServerAddress=192.168.0.2
;ServerTxt="Server2 name"

;[DebugLevel]
; 0 No printouts
; 1 Error printouts
; 2 Major events
; 3 Minor events
; 4 All printouts
;H323Debug=2
;UIDebug=2
;RTPDebug=2
;WAPDebug=2
;WEBDebug=2
;SIPDebug=1

[DefaultPacketSize]
DefaultMsPerPacket711=30   ;10,20 or 30ms. If not 30, then may
                           ;not be possible to call DBC 42x01 &
                           ;DBC 413 phones(that don't support
                           ;framesize negotiation).
DefaultMsPerPacket729=30   ;10,20,30,40,50 or 60ms. If not 30,
                           ;then may not be possible to call
                           ;DBC 42x01 & DBC 413 phones(that don't
                           ;support framesize negotiation).

[DIFFSERVTCP]
TrafficClass=2             ;Class B
DropPrecedence=2           ;Medium drop percentage

[DIFFSERVUDP]
TrafficClass=5        ;Expedited Forwarding
DropPrecedence=3      ;High drop percentage, recomended for EF

[Display]
ShowDTMFDigits=YES
ShowIPSettings=Enabled        ;Enabled/Disabled
AutoShowMonitorKeys=NO
BacklightMNSIncoming=YES
BlinkOPUMNSIncoming=YES
MissedCallAtBusy=YES
MissedCallTimer=0

;[Emergency]
;System1=MD110
;Address1=192.168.0.1
;Port1=1720
;System2=BP-R15
;Address2=192.168.0.1
;Port2=1720
;EmergencyNr=112
;EmergencyNr2=234
;EmergencyNr3=567
;EmergencyTxt1=”If the defualt text shall not be used”
;EmergencyTxt2=”Fire brigade X-city”
;EmergencyTxt3=”Fire brigade Y-city”
```

```
;ShowEmergencyTxt=NO
;A-Number=077601
;NumberingPlanOfA-Number=ISDN  ;ISDN/Private
;RouteId=PASSW

[FunctionKeysDBC434]
FreeOnSecond=0
CallBack=1
FollowMe=2
;Conference=
;Transfer=
;CallPickUp=
;CallWaiting=
;Intrusion=
EnablePBXStoring=NO

[FunctionKeysDBC444]
;RejectKey=ENABLED
;ENABLED/DISABLED default:ENABLED  DISABLED means that calls
can't be rejected

[FunctionKeysDBC446]
;RejectKey=ENABLED
;ENABLED/DISABLED default:ENABLED  DISABLED means that calls
can't be rejected

[HeadsetPreset]
;HeadsetTone=NO
;SpeakerRinger=YES

[IPPhoneAdministrator]
IPPhoneAdministrator=YES;Default = YES
;ServerAddress=192.168.0.1;Optional
;ServerPort=8080;Optional

[Language]
;Default language is English. Optional languages can be selected
LanguageFile=/dbc44x01/d44601-lang_R2B.txt
LanguageVersion=R2B
DBC444LanguageFile=/dbc44x01/d44401-lang_P1A1.txt
DBC444LanguageVersion=P1A1
;OptionalLanguage=EN,*08*0#
;OptionalLanguage=FR,*08*1#
;OptionalLanguage=DE,*08*2#
;OptionalLanguage=ES,*08*3#
;OptionalLanguage=IT,*08*4#
;OptionalLanguage=SV,*08*5#
;OptionalLanguage=FI,*08*6#
;OptionalLanguage=DA,*08*7#
;OptionalLanguage=NO,*08*8#
;OptionalLanguage=NL,*08*9#
;OptionalLanguage=PB,*08*9#
;OptionalLanguage=PL,*08*9#
;OptionalLanguage=PT,*08*9#
;OptionalLanguage=XL,*08*9#
;OptionalLanguage=SK,*08*9#
;OptionalLanguage=CS,*08*9#
;OptionalLanguage=ET,*08*9#
;OptionalLanguage=HU,*08*9#
;OptionalLanguage=LT,*08*9#
```

```
;OptionalLanguage=SL,*08*9#
;StartupLanguage=EN

[LEDs]
Cad0=185,5,5,5        ; LED flash 0 cadence, first on, first
                      ; off,second on, second off: in a call
Cad1=50,50,0,0        ; LED flash 1 cadence: call parked
Cad2=30,30,0,0        ; LED flash 2 cadence: incoming call

[LLDPMED]
LLDPMED=Enabled
TTL=120

[L2QOS]
PhonePort=6,1                    ;prio (0-7) ,VID(1-4094)
PCPort=0,2                       ;prio (0-7) ,VID(1-4094)
LANPort=0,0                      ;MAC based VLAN/802.1Q based
                                 ;VLAN , untagged/tagged PC
                                 ;originating traffic
BroadcastStormControlLimit=80    ;Limit for no. of broadcast
                                 ;packets received within 100ms
                                 ;time interval. Default is 80
                                 ;(800 b.c./s)
RememberVLAN=WARM                ;WARM = Remember VLAN ID only
                                 ;after warm reboot, COLD =
                                 ;Remember VLAN ID after both warm
                                 ;and cold reboot.

;[Messages]=NO
;VoiceMailNumber=

;[OMD]
;SNAddress=192.168.0.1
;SNPort=1700

[Password]
PasswordInputFormat=alphanumeric  ;alphanumeric/digit
PINorPassword=Password            ;PIN/Password
ServiceCodeChangePIN=*74*
;UseAdminPassword = DISABLED      ;ENABLED/DISABLED

;[RECORDER]
;LoggerIPAddress1=192.168.0.1     ;The telephone checks that the
                                  ;IP address from which the
                                  ;recording is ordered
                                  ;corresponds to the IP address
                                  ;in this parameter value.
;LoggerIPAddress2=192.168.0.2     ;See LoggerIPAddress 1 above.
;LoggerIPAddress3=192.168.0.3     ;See LoggerIPAddress 1 above.

; Record on demand setting when using a NICE system

;RODServerIPAddress=192.168.0.4 ;When record on demand (start
                                ;recording with function key)
                                ;shall be used, the ROD server
                                ;is mandatory.
;RODServerName=servername       ;The name of the BSF server. This
                                ;data must be provided by the
                                ;administrator for the NICE
                                ;Perform® recording system.
```

```
;RODServerCredentials=usr:pass ;BSFServerCredentials=
                               ;username:password. The login
                               ;credentials towards the BSF
                               ;server.
;RODServerGkId=1               ;This data is set and used in the
                               ;BSF server.

;Record on demand using recording equipment from ASC

;RODStart=http://192.168.0.1:8080/XVOIP/VOIPService?page=START&OPN=
;RODStop=http://192.168.0.1:8080/XVOIP/VOIPService?page=STOP&OPN=

; Record on demand setting when using another vendor

;RODStart=http://192.168.0.1/recbutton?command=start&user=
                               ;URL sent when pressing rec. key
;RODStop=http://192.168.0.1/recbutton?command=stop&user=
                               ;URL sent when pressing rec. key

;VoiceMailNumber=  ;Define a default Voice Mail Box Number

[RingCads]
Internal=20,100,0,1,0,0        ;Internal ring cadence
External=7,6,7,100,0,0         ;External ring cadence
Callback=6,8,6,8,0,0           ;Callback ring cadence
Extra=6,8,6,8,6,8              ;Extra ring cadence
Delay=7                        ;Delay for alerting in seconds
CallWaiting=RingSignal         ;RingSignal or Tone

;[RingLevel]

;MNSBusyRingLevel=NORMAL ;LOW = Low ring level on an incoming MNS
at active speech state. SILENCE = Silent ringing. Default is
NORMAL

[Security]
Security=DISABLED          ;ENABLED or DISABLED States if SRTP
                           ;& TLS shall be used
CertificateValidate=NO     ;States if the phone shall validate
                           ;the server certificate. If YES,
                           ;required that NTP-server parameter
                           ;is configured.
SecurityFallback=YES       ;states if it shall be allowed to
                           ;continue the registration in a not
                           ;secure way, if the TLS negotiation
                           ;fails.
SaveUserPassword=YES
SignallingEncrypted=YES    ;YES:the signalling is encrypted;
                           ;This is the default value. NO:NULL
                           ;cipher which means that no
                           ;encryption of the signalling is done
                           ;but the signalling sequence is the
                           ;same as when the signalling is
                           ;encrypted. This option can be used
                           ;at fault locating.
;RootCert=cacert.pem
UDPFilter=Enabled
UDPRateLimit=150           ;max 150 packets per 2 sec
TCPFilter=Enabled
```

```
;OpenTCPPort1=1720              ;Open traffic from/to TCP
destination port

;OpenUDPPort1=9200              ;Open traffic from/to UDP
destination port

;OpenUDPPort2=9200              ;Open traffic from/to UDP
destination port

;OpenUDPPort3=9200              ;Open traffic from/to UDP
destination port

[ServiceCodes]
CancelCallBack=#37#
CancelManualMessageWaiting=#31#

;[SIP]
;SIPDomain=MX3-lim1.terminallab.ebc.aastra.se
;SIPProxyAddress=130.100.26.166
;SIPProxyPort=5060
;SecureSIPProxyPort=5061
;SIPProxyAddress=192.168.0.1
;SIPProxyPort=5060
;SecureSIPProxyPort=5061
;SIPUseTelURI=NO
;SIPHookFlash=DTMF_INFO_!
;SIP_DTMF=INFO           ;RFC2833 or INFO

[STOREUSERDATA]
EnableStoring=NO
CompatibleWithD4=YES
;IPAddress=192.168.0.18
;FTPUname=Telephone
;FTPPword=Telephone
;UseSFTP=NO

[SuffixServices]
Intrusion=4
CallWaiting=5
CallBack=6
CallPickUp=8
Conference=3

[Time]
TIMEZONE=CET:60:0:032802:103102 ;Summer time, Starts 28
                                ;Mars 2 pm, Ends 31 October 2 pm.
;NTP-server=192.168.0.1

[Tones]
Dial tone=
0,0,                     ; One cadence, One frequency
500,0,0,0,0,0,           ; 500ms on, 0ms off, 0ms on, 0ms off,
                         ; 0ms on, 0ms off
425,-10,0,0,0,0,         ; 425Hz, -10dbm0, 0Hz, 0dBm0, 0Hz,
                         ; 0dBm0,
0,0                      ; Tone in Voice is off, sine wave

Special dial tone=
1,0,                     ; One cadence, One frequency
500,50,0,0,0,0,          ; 500ms on, 50ms off, 0ms on, 0ms off,
                         ; 0ms on, 0ms off
```

```
425,-10,0,0,0,0,        ; 425Hz, -10dbm0, 0Hz, 0dBm0, 0Hz, 0dBm0
0,0                     ; Tone in Voice is off, sine wave

Busy tone=
1,0,                    ; One cadence, One frequency
500,500,0,0,0,0,        ; 500ms on, 500ms off, 0ms on, 0ms off,
                        ; 0ms on, 0ms off
425,-10,0,0,0,0,        ; 425Hz, -10dbm0, 0Hz, 0dBm0, 0Hz, 0dBm0
0,0                     ; Tone in Voice is off, sine wave

Alerting tone=
1,0,                    ; One cadence, One frequency
1000,4000,0,0,0,0,      ; 1000ms on, 4000ms off, 0ms on, 0ms
                        ; off, 0ms on, 0ms off
425,-10,0,0,0,0,        ; 425Hz, -10dbm0, 0Hz, 0dBm0, 0Hz, 0dBm0
0,0                     ; Tone in Voice is off, sine wave

Congestion tone=
1,0,                    ; One cadence, One frequency
200,200,0,0,0,0,        ; 200ms on, 200ms off, 0ms on, 0ms off,
                        ; 0ms on, 0ms off
425,-10,0,0,0,0,        ; 425Hz, -10dbm0, 0Hz, 0dBm0, 0Hz, 0dBm0
0,0                     ; Tone in Voice is off, sine wave

Special information tone=
3,4,                    ; Three cadences, Three frequencies
333,1,333,1,333,1000,   ; 333ms on, 0ms off, 333ms on, 0ms off,
                        ; 333ms on, 1000ms off
950,-10,1400,-10,1800,-10,; 950Hz, -10dbm0, 1400Hz, -10dBm0,
                        ; 1800Hz, -10dBm0
0,0                     ; Tone in Voice is off, sine wave

Call waiting tone=
2,0,                    ; Two cadences, One frequency
200,600,200,3000,0,0,   ; 200ms on, 600ms off, 200ms on, 3000ms
                        ; off, 0ms on, 0ms off
425,-10,0,0,0,0,        ; 425Hz, -10dbm0, 0Hz, 0dBm0, 0Hz, 0dBm0
0,0                     ; Tone in Voice is off, sine wave

Offhook queuing tone=   ;also used as verification tone for
                        ; BP-R15
1,0,                    ; One cadence, One frequency
2000,990,0,0,0,0,       ; 2000ms on, 990ms off, 0ms on, 0ms off,
                        ; 0ms on, 0ms off
1400,-10,0,0,0,0,       ; 1400Hz, -10dbm0, 0Hz, 0dBm0, 0Hz,
                        ; 0dBm0
0,0                     ; Tone in Voice is off, sine wave

On hold tone=           ; On hold tone
2 ,1,                   ; Two cadences, Two frequencies
10,150,10,5000,0,0,     ; 10ms on, 150ms off, 10ms on, 5000ms
                        ; off, 0ms on, 0ms off
400,-10,300,-10,0,0,    ; 400Hz, -10dbm0, 300Hz, -10dBm0, 0Hz,
                        ; 0dBm0
0,0                     ; Tone in Voice is off , sine wave

External dial tone=
0,0,                    ; One cadence, One frequency
500,0,0,0,0,0,          ; 500ms on, 0ms off, 0ms on, 0ms off,
                        ; 0ms on, 0ms off
425,-10,0,0,0,0,        ; 425Hz, -10dbm0, 0Hz, 0dBm0, 0Hz,
```

```
                        ; 0dBm0,
0,0                     ; Tone in Voice is off , sine wave

Waiting voice tone=
1,0,                    ; One cadence, One frequency
30,330,0,0,0,0,         ; 30ms on, 330ms off,
425,-10,0,0,0,0,        ; 425Hz, -10dbm0,
0,0                     ; Tone in Voice is off , sine wave

Headset ring tone=
1,0,                    ; One cadence, One frequency
200,200,200,0,0,0,      ; 200ms on, 200ms off, 200ms on, 0ms
                        ; on, 0ms off
425,-10,0,0,0,0,        ; 425Hz, -10dbm0, 0Hz, 0dBm0, 0Hz, 0dBm0
0,0                     ; Tone in Voice is off, sine wave

Recording tone=         ; Two cadences, One frequency
0,0,                    ; 75ms on, 50ms off, 50ms on, 14720ms
                        ; off, 0ms on, 0ms off
75,50,50,14720,0,0,     ; 75ms on, 50ms off, 50ms on, 14720ms
                        ; off, 0ms on, 0ms off
1400,-310,0,0,0,0,      ; 1400Hz, -31dbm0, 0Hz, 0dBm0, 0Hz,
                        ; 0dBm0
0,0                     ; Tone in Voice is off, sine wave

[WAP]
MaxAttempts=3;2-8
RetransPeriod=2000;2000-7000
CountryCode=46;The code of the country where the
;PBX is located
RouteAccessNumber=00;The route access number
CountryCodePrefix=00;The prefix of the code of the country
;where the PBX is located
AreaCodePrefix=0;The prefix of the area code in the
;country where the PBX is located
AreaCode=8;The area code in the country where the
;PBX is located

;[WEBBrowser]
;HomeAddress=http://www.aastra.com;Home address
;DirectoryAddress=192.168.0.1/mobileexec;Corporate Directory
address
;DBC444DirectoryAddress=192.168.0.1/CorpDir/d4/d4.asxp
;WebProxyIP=192.168.0.1  ;Proxy IP address
;WebHttpPort=8080        ;Proxy port number
;WebHttpsPort=
;WebDNS1=192.168.0.1     ;IP address of the first DNS server
;WebDNS2=192.168.0.1     ;If the first DNS server does not work,
                         ;use the second one
;NoProxyAddress=internal.aastra.com,192.168.0.1
                         ;Bypass list. The address of the web
                         ;sites within the intranet. The address
                         ;are separated by commas.
;UserAgent=DBC44601      ;User Agent of the browser. Corporate
                         ;Directory server uses it to identify
                         ;the D5 phones.
;DBC444UserAgent=DBC44401
;UseWEBBforCorpDir=NO
```

```
;[WebServer]
;webInterfaceForUser=ENABLED        ;ENABLED,
                                    ;ENABLED_WITH_DEFAULT_PWD
                                    ;or DISABLED,  ENABLED as
                                    ;Default
```

## 4.2 DBC 43X 01 USED WITH BUSINESSPHONE

In this example DBC 433 and DBC 434 phones are used with BusinessPhone R16 or later.

```
[System]
;------------------------------------------------------------
; ; File: d43x01-config.txt
; Product id and version:  CAA 158 0064 R1A
; Release date 2008-05-30
;------------------------------------------------------------
; The system may have the  values MD110, MX-ONE, BP-R16 or MD-E.
;System=MD110
;System=MX-ONE
System=BP-R16
;System=MD-E
OverLap=YES
RRQTtl=OFF
;GatekeeperID=LIM*
;Domain=aastra.se
;PrimaryGKAddress=192.88.30.200
;SecondaryGKAddress=192.88.30.201
GatekeeperDiscovery=NO
AdminPassword=RbQdz9yS
;LogOffRestriction=LogOffAllowed
;LogOffRestriction=DefaultNumberUsed
;LogOffRestriction=PermitIndividualLogOn
;LogOffTime=03:00
OperatorNumber=09

[Software]
;--------------------------------------------------------
; The following files are mandatory:
; Application File: d43x01-applic_Rev.dat   (CAA158 0062)
; Config File:      d43x01-config.txt       (CAA158 0064)
; Language File     d43x01-lang_Rev.txt     (CAA158 0063)
It can happen that the superboot has to be updated:
Superboot           d43x01-sboot_Rev.hex    (CAA158 0061)
;--------------------------------------------------------
ApplicationFile=/dbc43x01/d43x01-applic_R1A.dat
ApplicationVersion=R1A

[802.1x]

[Absence]
;FollowMe=*21*extensionNo#,#21#
;ExtFollowMe=*22#extensionNo#,#22#
;AbsenceReason=*23*reasonCode#,#23#
;Lunch=0*hhmm
;Busy=1
;Absent=2*hhmm
```

```
;Meeting=3*hhmm
;Trip=4*mmdd
;Course=5*mmdd
;Vacation=6*mmdd
;Day off=7*mmdd
;GoneHome=8*hhmm
;Illness=9*hhmm
;Special=9*mmdd
;ParentalLeave=;
;OfficialMatter=7*mmdd
;FreeTime =6*mmdd

[Authorization] ;replaces the following code with "-"
Code=*61*        ;Account code for Standard Appl. System
Code=*72*        ;Auth. code for Standard  Appl. System
Code=*73*        ;Lock common authorization code
Code=#73*        ;Unlock unlock common authorization code
Code=*74*        ;Change individual  authorization code
Code=*75*-*n#    ;Dial with individual authorization code  n
                 ;means that the No. will be visible
Code=*76*        ;Lock individual  authorization code
Code=#76*        ;Unlock individual authorization code

[AutoNeg]
AutoNegLANPort=YES    ;When AutoNeg is set to YES, Speed and
                      ;Duplex will not be read.
;SpeedLANPort=100     ;LAN transmit/receive speed =  100 Mbit/s
;DuplexLANPort=FULL   ;Full duplex,
AutoNegPCport=YES
;SpeedPCPort=100
;DuplexPCPort=FULL

;[BackupGK]
;System=MD-E
;IPAddress=153.88.30.203

[Codecs]
;G.722       ;Not allowed for MX-ONE TSW. Works only
             ;for IP to IP calls in H.323 mode
;G.711 is mandatory but the priority order is  optional
G.711A
G.711U
GSMEFR
G.729A
G.729AB
G.723

[DefaultPacketSize]
DefaultMsPacket711=30
DefaultMsPacket729=30

[DIFFSERVTCP]
TrafficClass=2        ;Class B
DropPrecedence=2      ;Medium  Drop Precedence

[DIFFSERVUDP]
TrafficClass=5        ;Expedited forwarding
DropPrecedence=3      ;High Drop Precedence
```

```
[Display]
ShowDTMFDigits=YES
ShowIPSettings=ENABLED

;[Emergency]
;System1=MD110
;Address1=192.100.26.154
;Port1=1720
;System2=BP-R15
;Address2=192.100.26.156
;Port2=1720
;EmergencyNr=112
;A-Number=077601
;NumberingPlanOfA-Number=ISDN
;RouteId=PASSWORD

;[FunctionKeysDBC434]
;FreeOnSecond=0
;CallBack=1;
;FollowMe=2
;Transfer=
;CallPickUp=
;CallWaiting=
;Intrusion=
;EnablePBXStoring=NO

[HeadsetPreset]
;HeadsetTone=NO
;SpeakerRinger=YES

[FunctionKeysDBC444]
;RejectKey=ENABLED;ENABLED/DISABLED default:ENABLED DISABLED
means that calls can't be rejected

[FunctionKeysDBC446]
;RejectKey=ENABLED ;ENABLED/DISABLED default:ENABLED DISABLED
means that calls can't be rejected

[IPPhoneAdministrator]
IPPhoneAdministrator=YES
;ServerAddress=192.100.17.70
;ServerPort=8080

[Language]
; Default language is English. Optional languages can be
; selected e.g. Swedish, Spanish
LanguageFile=/dbc43x01/d43x01-lang_R1A.txt
LanguageVersion=R1A OptionalLanguage=ES,*08*04#
StartupLanguage=EN

[LEDs]
Cad0=185,5,5,5   ; LED flash cadence 0, first on, first off,
                 ; second on, second off
Cad1=50,50,0,0   ; LED flash cadence 1, 500ms, 500ms,  0ms, 0ms
Cad2=30,30,0,0   ; LED flash cadence 2

[LLDPMED]
LLDPMED=Enabled
TTL=120
```

```
[L2QOS]
PhonePort=6,1          ;Priority, VLAN id (1-4094).
PCPort=0,2             ;User priority, VLAN id (1-4094)
LANPort=0,0            ;0 = No VLAN (untagged), 0 =  untagged
BroadcastStormControlLimit=8

[OMD] ;
;SNAddress=130.100.188.111
;SNPort=1700

[Password]
ServiceCodeChangePIN=*74*
PINorPassword=Password
;PasswordInputformat=alphanumeric
;UseAdminPassword=DISABLED

;[RECORDER]
;LoggerIPAddress1=192.168.0.1
;LoggerIPAddress2=192.168.0.2
;LoggerIPAddress3=192.168.0.3

; Record on demand settings when using a NICE system

;RODServerIPAddress=192.168.0.4
;RODServerName=servername
;RODServerCredentials=usr:pass;BSFServerCredentials=
;RODServerGkId=1

; Record on demand settings when using another vendor

;RODStart=http://192.168.0.1/recbutton?command=start&user=67609
;RODStop=http://192.168.0.1/recbutton?command=stop&user=67609

RecordingTone=ENABLED  ;ENABLED or DISABLED, default is DISABLED

[Ringcads]
Internal=20,100,0,1,0,0  ; Internal ring cadence
External=7,6,7,100,0,0   ; External ring cadence
Callback=6,8,56,8,0,0    ; Callback ring cadence
Extra=6,8,6,8,6,8        ; Extra ring cadence
Delay=7                  ; Delay for alerting, in seconds
CallWaiting=RingSignal   ;RingSignal or Tone

[Security]
;Security=DISABLED
;CertificateValidate=NO
;SecurityFallback=YES
;SaveUserPassword=YES
;RootCert=cacert.pem
UDPFilter=Enabled
UDPRateLimit=150
TCPFilter=Enabled

;OpenTCPPort1=1720 ;Open traffic from/to TCP destination port
;OpenUDPPort1=9200 ;Open traffic from/to UDP destination port
;OpenUDPPort2=9200 ;Open traffic from/to UDP destination port
;OpenUDPPort3=9200 ;Open traffic from/to UDP destination port

[STOREUSERDATA]
EnableStoring=NO
```

```
CompatibleWithD4=YES
;IPAddress=192.168.0.18
;FTPUname=Telephone
;FTPPword=Telephone

[SuffixServices]
Intrusion=4
Callwaiting=5
CallBack=6
CallPickup=8
CancelCallBack=#37#

;[Time]
;TIMEZONE=CET:60:0:04:01:04:11:01:01
;NTP-server=192.100.12.8

;[Tones]
;The default values for the tones are used when no data
identifiers for the header are defined

[WAP]
MaxAttempts=3
RetransPeriod=2000
;CountryCode=46
;RouteAccessNumber=00
;CountryCodePrefix=00      ;The prefix of the code  of the
                           ;country where the PBX is located
;AreaCodePrefix=0          ;The prefix of  the area code in the
                           ;country where the PBX is ;located
;AreaCode=8

[WebServer]
webInterfaceForUser=Enabled
```