R1000-, R3000-, R4000-Serie

Release Notes

System Software 7.6.6

Version 1.0

Purpose This document describes new features, changes, and solved problems of **System Software 7.6.6.**

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How to reach Funkwerk Enterprise Communications GmbH

Funkwerk Enterprise Communications GmbH Suedwestpark 94 D-90449 Nuremberg Germany Telephone: +49 180 300 9191 0 Fax: +49 180 300 9193 0 Internet: www.funkwerk-ec.com	Funkwerk Enterprise Communications 6 Avenue de la Grande Lande - CS 20102 33173 Gradignan cedex France Telephone: +33 (0)1 61 37 32 76 Fax: +33 (0)1 61 38 15 51 Internet: www.funkwerk-ec.com

1 Important Information

Please read the following information about System software 7.6.6 carefully to avoid problems when updating or using the software.

1.1 Applicability

System software 7.6.6 is available only for the following devices and cannot be used on other devices:

- **R1200**
- **R1200w**
- **R3000**
- **R3000w**
- **R3400**
- **R3800**
- **R4100**
- **R4300**

1.2 Incompatibility

Configurations created or saved with **System software 7.6.6** are incompatible with some versions of our system software.

Take note, however, of the following indications regarding the update and the possibilities of a downgrade.

1.2.1 Preparation and update

To prepare and carry out an update to **System software 7.6.6**, proceed as follows:

1. Backup the current boot configuration. Use one of the following possibilities:
   a) In the SNMP shell, enter `cmd=save path=boot.alt`. This backs up the current boot configuration in the flash ROM of your gateway under the name "boot.alt".
   b) On a computer on your LAN, start a TFTP server and export the current boot configuration via the ***Configuration Management*** menu of the Setup Tool. To do this, select:
   – ***Operation*** = *put (FLASH -> TFTP)*
   – ***TFTP server IP address*** = *<IP address of the TFTP servers on the LAN>*
   – ***TFTP File name*** = *boot.alt*
   – ***Name in Flash*** = *boot*
2. Carry out the update to **System software 7.6.6** as usual and restart the gateway.
   The gateway will start with the new software, the existing boot configuration will be used.

1.2.2 Downgrade

If you wish to carry out a downgrade, proceed as follows:

1. Replace the current boot configuration with the previous backup version. Use one of the following possibilities:
   a) In the SNMP shell, enter `cmd=move path=boot.alt pathnew=boot`. This overwrites the current boot configuration with the previous backup version. The configuration named "boot.alt" is thereby deleted from the flash ROM (if you wish to keep this in flash, use `cmd=copy` instead of `cmd=move`).
   b) On a computer on your LAN, start a TFTP server and import the previously backed up configuration via the ***Configuration Management*** menu of the Setup Tool. To do this, select:

- **OPERATION** = get (TFTP -> FLASH)
- **TFTP SERVER IP ADDRESS** = *<IP address of the TFTP servers on the LAN>*
- **TFTP FILE NAME** = *boot.alt*
- **NAME IN FLASH** = *boot*

2. Carry out the downgrade to the desired software version.
3. Reboot the gateway. The device will start with the previously backed up boot configuration and the old version of the system software.

1.3 Update for VoIP

Since you wish to use the new Voice over IP (VoIP) functionality of our **System software 7.6.6**, you may, depending on the type, serial number and a further parameter of your device, the so-called *Ident Version*, need to update this *Ident Version*. The update is necessary so that the DSP module is correctly recognised.

The following devices will need an update if digits four and five of the devices serial number constitute a number less than 17.

- **R1200**
- **R1200w**
- **R3000**
- **R3000w**
- **R4100.**

The update applies only to devices for which the value of the *Ident Version* Parameter is less than 1.8 or 1.71.

The value 1.8 applies for devices

- **R1200**
- **R1200w**
- **R3000**
- **R3000w.**

The value 1.71 applies for the **R4100** gateway.

To find out whether your device needs an update, first check the serial number. If appropriate, then check the *Ident Version* parameter. If needed, then proceed with the update of the *Ident Version*.

Proceed as follows:

1. Check the serial number of your device. You will find this on the sticker on the underside of the case, beneath the designation *Serial Number*, e.g. *R1D160006100009*.
2. Check digits four and five of this number. If they constitute a number less than 17 (as in the above example), check the *Ident Version* of your device. If they constitute a number equal to 17 or more, no update of the *Ident Version* is needed.
3. To check the *Ident Version*, log into the gateway with a serial connection. Enter `show rev` in the SNMP shell.
   You will see a value for the *Ident* parameter, e.g. *V.1.7*.
   Since 1.7 is less than both 1.71 and 1.8, the condition for an update of the *Ident Version* is in this case fulfilled for all devices concerned. If the condition is not fulfilled, no update of the *Ident Version* is needed.

The update of the *Ident Version* is carried out in the BOOTmonitor with a BLUP (Bintec Large Update).

Attention!

The Ident Version update carries the risk that if the update of one of the components fails, as a result of a power supply interruption, for example, your gateway can no longer be started. In this case you must return the gateway to your dealer.

Proceed as follows:

1. Configure a computer on your local network as a TFTP server. On a Windows-PC, you can use the DIME Tools TFTP server for this (the DIME Tools are to be found on the companion CD):
2. Copy the BLUP file
   (e.g. bl_r1200_r3000_r4100_ident_update.rny from our Web-server) to the TFTP directory of the TFTP servers on your local network.

3. Reboot the gateway by entering `cmd=reboot` in the SNMP shell. Your gateway will reboot and, after a series of log messages, you will see the following prompt:

```
Press <sp> for boot monitor or any other key to boot
system
```

4. Now press the **space bar** within four seconds to enter BOOTmonitor mode (all values shown here are only examples):

```
R1200 Bootmonitor V.7.5 Rev. 1 from 2007/04/17 00:00:00
Copyright (c) 1996-2007 by Funkwerk Enterprise Communications GmbH

(1) Boot system
(2) Software update via TFTP
(3) Software update via XMODEM
(4) Delete configuration
(5) Default BOOTmonitor parameters
(6) Show System Information

Your Choice>
```

5. Select `2` and then press **Enter**. You must now enter the IP address of the gateways, the IP address of the TFTP servers and the name of the file to be loaded (in the case of the BLUP, bl_r1200_r3000_r4100_ident_update.rny). After each, press **Enter**:

```
Your choice> 2
Enter local IP address [192.168.1.254]:
Enter IP address of TFTP server [192.168.1.1]:
Enter file name of image [b6105.x8a]:
bl_r1200_r3000_r4100_ident_update.rny

Are your entries correct (y or n) ?
```

6. Check the entries. If they are correct, press `y` and then **Enter**:

```
Starting file transfer
......................................................................
........OK (553172 bytes received)
Checking new image ... OK

Loaded new image has release 7.5.1.100

Now choose from the following:

(u) Update Flash ROM
(r) Write image to RAM and start it
(e) Exit

Enter (u, r or e):
```

7. To update the software, press `r` and then **Enter**:

```
Booting BOSS...
boss image started at 0x18d0034



R1200 BLUP V.7.5 Rev. 1 from 2007/04/17 00:00:00
Copyright (c) 1996-2007 by Funkwerk Enterprise Communications GmbH

List of files in this update (len 393372):
  Version    Length  Name
      1.8    131124  Ident
      1.8    131124  Ident
     1.71    131124  Ident

Proceed with update (y or n) ?
```

8. Press `y` to update all necessary files and write them to the flash memory:

```
*** Don't power-off your router while the update takes place ***


Updating Ident
Erasing Flash-ROM . OK
Writing Flash-ROM . OK
Verify Flash ROM . OK

Updating Ident
 *** Ident image not matchin HW (2) [skipped]

Updating Ident
 *** Ident image not matchin HW (2) [skipped]
Rebooting...

*** R1200 (Hardware-rev. 1.0, Firmware-Rev. 1.0) ***

CPU   Check ... passed (MPC 8272 @ 400(100.0 MHz)
SDRAM Check .............     passed (32 MByte)
FLASH Check ........ passed (8 MByte)

*** Selftest passed ***
```

After the reboot, the update is complete and you can use Voice over IP.

2 New Feature

From System software 7.6.6, you have an extension of the Voice-over-IP implementation available, providing a significantly extended range of functions for telephoning over the Internet.

2.1 VoIP

System software 7.6.6 supports the function of a media gateway. A media gateway serves as a translation instance between different telecommunications networks, e.g between the plain old phone network and the next generation networks (IP networks).

With the Funkwerk Media Gateway, a company equipped with an automatic PBX on a wired telephone network can be connected to a SIP Trunking Service Provider on the Internet in order to use IP telephony.

The Funkwerk Media Gateway supports the binding of several SIP Provider Accounts. With this gateway, you can set up extensions, create an extension number plan and configure exchange functions and optimise voice data transmission for low bandwidth of the upload connection.

Note

Your device must be fitted with a DSP module to be able to use the media gateway functions. Information on building in the DSP module is provided in the installation instructions included with the module.

In the Setup Tool, the ***VoIP*** contains the following submenus and fields:

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP]: Configuration                                        MyGateway

                 Application Level Gateways >

                 Accounts

                 Extensions

                 PBX Configuration >



                 Dynamic Bandwidth Control >

                      EXIT

```

Application Level Gateways

To enable IP telephones to connect by SIP to a VoIP Provider your device has an Application Level Gateway (ALG), i.e. an appropriate proxy that implements the necessary NAPT and firewall releases.

Note

The Application Level Gateway must always be used if NAT is enabled on the interface that makes the connection to the Internet.

In the ***VoIP*** ➜ ***APPLICATION LEVEL GATEWAYS*** menu, a list is shown of all configured Application Level Gateway entries. These entries enable the ALG. Each entry defines a particular TCP or UDP destination port that is to be supervised by the ALG. As normally delivered, there are two entries configured for the SIP Ports TCP 5060 and UDP 5060 in accordance with the IANA definition.

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP][ALG]: Application Level Gateway configuration         MyGateway


 Description      Type   Status    Port   Protocol    LLT
 -----------------------------------------------------------
 SIP TCP 5060     SIP    disable   5060      tcp       off
 SIP UDP 5060     SIP    disable   5060      udp       on




 -----------------------------------------------------------

        SIP Endpoint Configuration >



    ADD                 DELETE              EXIT

```

In the ***VoIP*** ➜ ***Application Level Gateways*** ➜ ***ADD/EDIT*** menu, you can configure an Application Level Gateway entry.

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP][ALG][ADD]: Application Level Gateway                  MyGateway

   Application Level Gateway settings:

          Description                NEW Gateway
          Proxy Type                 SIP

          Adminstatus                enable
          Destination Port           9999
          Protocol                   udp
          Session timeout            7200

          Low Latency Transmission   on




          SAVE                                CANCEL

```

The ***VoIP* ➜ *Application Level Gateways* ➜ *ADD/EDIT*** menu, consists of the following fields:

Field	Value
Description	Here you give the Application Level Gateway entry a name.
Proxy Type	Defines the protocol to be supervised by the proxy. Possible values: ■ *SIP* (default value): Session Initiation Protocol.
Adminstatus	Defines if the proxy is to be active. Possible values: ■ *enable* (default value): The proxy is enabled. ■ *disable*: The proxy is disabled.
Destination Port	Here you enter the port to be supervised by the proxy. For each destination port to which VoIP clients from the LAN can connect, you must configure a proxy. The ports can be provider-specific. The default value is *5060*.
Protocol	Defines the protocol to be used. Possible values: ■ *udp* (default value): User Datagram Protocol ■ *tcp*: Transmission Control Protocol.

Field	Value
Session timeout	Time in seconds for which a session stays up if no data packets are sent or received. This value must be greater than the SIP Expire Time of the connected SIP client (SIP telephone, terminal adapter etc.) Default value: *7200*.
Low Latency Transmission	Mechanism to minimise the transit time of VoIP data packets between two subscribers. This guarantees good voice quality with high line load. Note that Low Latency Transmission does not have to be switched on if the media gateway terminates the VoIP connections. Possible values: ■ *on* (default value): voice quality is optimised. ■ *off*: voice quality is not optimised.

Table 2-1: Fields in the ***VoIP* ➜ *Application Level Gateways* ➜ *ADD/EDIT*** menu

Under ***VoIP* ➜ *Application Level Gateways* ➜ *SIP Endpoint Configuration***, the SIP sessions currently being supervised by the ALG are shown. With **ADD**, you can add static entries for SIP servers / proxies in the LAN that are to be accessible from the WAN across the NAPT barrier.

Note

All automatically generated entries that are not used for longer than 24 hours are also automatically deleted from the table under ***VoIP* ➜ *Application Level Gateways* ➜ *SIP Endpoint Configuration***.

Only if SIP terminals from outside are to register with an internal SIP server must you enter values for the SIP server yourself ***VoIP ➜ Application Level Gateways ➜ SIP Endpoint Configuration ➜ ADD/EDIT***.

```
R1200w Setup Tool               Funkwerk Enterprise Communications GmbH
[VOIP][ALG][ADD]: Terminal Endpoint                           MyGateway



       Endpoint Type                 client
       Protocol                      udp
       Internal Address
       Remote Port                   0
       External Port                 0



                SAVE                           CANCEL

```

The menu contains the following fields:

Parameter	Value
Endpoint Type	SIP terminal that is used. Possible values: ■ *client* (default value): Used for SIP clients, e.g. telephones. ■ *server*: Used for SIP servers in the LAN so that SIP clients from the WAN can register.
Protocol	Protocol used for data transmission. Possible values: ■ *udp* (default value): User Datagram Protocol ■ *tcp*: Transmission Control Protocol If a protocol has been automatically recognised, it should not be changed.
Internal Address	Internal IP address of the SIP terminal, e.g. of a SIP phone. The SIP terminal uses this IP address to register at the provider.
Remote Port	Only for ***ENDPOINT TYPE*** = *client*. Port number which is to be used to connect to the provider; the provider listens at this port. SIP default value: *5060*.
Internal Port	Only for ***ENDPOINT TYPE*** = *server*. Port number, if a server is used in the own network. SIP default value: *5060*.

Parameter	Value
External Port	External port that is to be bound to the internal port. For clients, the external port is recognised automatically and should not be changed.

Table 2-2: Fields in the ***VoIP*** ➜ ***Application Level Gateways*** ➜ ***SIP Endpoint Configuration*** ➜ ***ADD/EDIT*** menu

Accounts

If your want your device is to connect to other SIP servers (e.g. servers of Internet SIP Service providers), you can configure the necessary entries here. In this case, the media gateway acts as a SIP client.
Furthermore, you can configure the entries for SIP trunking scenarios here. In this case, the media gateway acts as a SIP server for other SIP servers. An example for this is the connection of a SIP PBX (e.g. Asterisk) to the media gateway.
This means that not only all SIP provider accounts are configured here but also direct dial-in PBXs connected with the media gateway.

Note

In no case should you use this menu to configure extensions, i.e. for SIP clients or PSTN clients such as SIP telephones, terminal adapters or ISDN telephones

Extensions can be configured in the ***VoIP*** ➜ ***Extensions*** ➜ ***ADD/EDIT*** menu.

The desired accounts are configured in the ***VOIP*** ➜ ***ACCOUNTS*** ➜ ***ADD/EDIT*** menu. Here you configure all necessary fields for the registration, independent of mode (client or server).

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP][PROV][ADD]: Account Configuration                     MyGateway

       Description

       Admin State               enabled
       Oper State                down

       Domain / Proxy
       User ID
       Password / PIN

       Port Number               5060

       Assigned VoIP Protocol    SIP

       Advanced Settings >

                SAVE                         CANCEL
```

The menu contains the following fields:

Parameter	Value
Description	Here you give the account a name. Maximum number of characters: *40*.
Admin State	Determines the administration state of the accounts. Possible values: ■ *enabled* (default value): the account is used. ■ *disabled*: the account is not used.
Oper State	Shows the current operating state of the accounts. Possible values: ■ *down*: not registered. ■ *trying*: registration process in progress. ■ *up*: registered. ■ *failed*: registration process failed. ■ *blocked*: the account blocked. ■ *disable*: the administration state (***ADMIN STATE*** field) is *disabled*.
Domain / Proxy	Here you enter the IP address or domain name (FQDN) of the remote SIP registrar or SIP proxy server. Maximum number of characters: *40*.
User ID	In SIP client mode, you enter the username for authentication here if your VoIP provider has assigned one for you. In SIP server mode, you must set the user-name. Maximum number of characters: *40*.

Parameter	Value
Password / PIN	In SIP client mode, the VoIP provider gives you a ***PIN*** or ***PASSWORD*** for authentication. You must enter this value here. In SIP server mode, you must set this value. Maximum number of characters: *40*.
Port Number	Number of the TCP or UDP port to be used for the connection. Default value: *5060*.
Assigned VoIP Protocol	Protocol to be used for the connection. Possible values: ■ *SIP*(default value): Uses the SIP protocol.

Table 2-3: Fields in the ***VOIP*** ➜ ***ACCOUNTS*** ➜ ***ADD/EDIT*** menu

Under ***VOIP*** ➜ ***ACCOUNTS*** ➜ ***ADD/EDIT*** ➜ ***ADVANCED SETTINGS***, make the settings for the SIP protocol and other specific settings.

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP][PROV][ADD][SIP]: Advanced Settings                    MyGateway


       Outbound Proxy

       Auth ID
       Realm

       Called Address                auto
       Registration Mode             on
       Expire Time                   300

       Transport                     UDP

       DDI Settings >
       Codec Settings >


                SAVE                          CANCEL

```

The submenu contains the following fields:

Parameter	Value
Outbound Proxy	Name or IP address of the SIP outbound proxy server. Maximum number of characters: *32*. Here you must enter a name or an IP address only if, for all SIP sessions via this account, the communication is not to be direct but via a further proxy. Enter a name or IP address only if this is explicitly specified by the provider.
Auth ID	Authentication Identifier Here you can enter a name that is to be used for authentication at the outbound proxy server. If you do not enter a name, the name in the ***User ID*** field is used. Enter a name only if this is explicitly specified by the provider.
Realm	Here you can enter an additional domain name or a further IP address of the SIP Proxy server. If you do not enter a domain name or an IP address, the value in the ***Domain / Proxy*** field is used. Enter a name or an IP address only if this is explicitly specified by the provider.

Parameter	Value
Called Address	Defines where and how the destination address is to be transferred. Note your provider's instructions. Possible values: ■ *auto* (default value): The "Display field is checked first, if empty, the destination address is written into the "User" field. ■ *user*: The destination address is transferred in the "User" field of the SIP header. ■ *display*: The destination address is transferred in the "Display" field of the SIP header.
Registration Mode	Enables or disables the SIP REGISTER registration mechanism. Normally, every SIP client (user) sends its current position to a REGISTRAR server by means of a REGISTER message. This information about the user and his current address is held by the REGISTRAR server and queried by other proxies to find the user. Apart from this standard procedure, the relevant data can also be sent to a particular IP address that is already known to the correspondent. Registration and authentication are not then needed and the setting ***Registration Mode*** = *off* must be made in this case. An example of this method is Microsoft Exchange SIP. Possible values: ■ *on* (default value): the registration mechanism is enabled. ■ *off*: the registration mechanism is disabled.

Parameter	Value
Expire Time	Time in seconds after which the current registration becomes invalid and a new registration request is therefore sent. Possible values: *0* .. *99999*. Default value: *300*. In answer to a REGISTER request, a server can set another Expire Time which overwrites the setting here.
Transport	Protocol to be used for data transport. Possible values: ■ *UDP* (default value): User Datagram Protocol ■ *TCP*: Transmission Control Protocol

Table 2-4: Fields in the ***VoIP ➜ Accounts ➜ ADD/EDIT ➜ Advanced Settings*** submenu

DDI In the ***VoIP ➜ Accounts ➜ ADD/EDIT ➜ Advanced Settings ➜ DDI Settings*** menu, you can make the settings for Direct Dial-In. An incoming call can be routed to just one terminal device (direct dial-in). For an outgoing call, the caller can be indicated to the called party.

```
R1200w Setup Tool                  Funkwerk Enterprise Communications GmbH
[VOIP][PROV][ADD][SIP][DDI]: DDI Settings                        MyGateway

      DDI Mode                    server

      CLI transfer Mode           display and Username

      Subscriber Number



            SAVE                              CANCEL
```

The menu contains the following fields:

Parameter	Value
DDI Mode	Defines in which Direct Dial-In mode the media gateway is operated. Which value can be used depends from the provider. Possible values: ■ *server*: the media gateway is operated as SIP server. ■ *client*: the media gateway is operated as SIP client. ■ *gateway trunk*: this setting is used to connect a Swyx software based IP PBX. ■ *off* (default value): the DDI mode is not used.

Parameter	Value
CLI transfer Mode	Only for ***DDI Mode*** = *server* and for ***DDI Mode*** = *client*. Defines the position of the DDI sender (caller) ID for outgoing calls. (For incoming calls, the subscriber number is deduced automatically from the SIP header.) Possible values: ■ *display only*: the sender Id is placed in the "Display" field of the SIP header. ■ *username only*: the sender Id is placed in the "User" field of the SIP header. ■ *display and username*: the sender Id is placed in both "Display" and "User" fields of the SIP header. ■ *P-Preferred-Identity*: The so-called "p-preferred-identity" field is added to the SIP header and contains the sender Id. ■ *P-Asserted-Identity*: The so-called "p-asserted-identity" field is added to the SIP header and contains the sender Id. ■ *disable*: the DDI sender Id is not sent.
Subscriber Number	Only for ***DDI Mode*** = *server*. Here you can set a number that is added as a prefix to the sender number for outgoing calls and is removed from the sender address for incoming calls. This corresponds to the trunk (exchange) number of an exchange.

Table 2-5: Fields in the ***VoIP*** ➜ ***Accounts*** ➜ ***ADD/EDIT*** ➜ ***Advanced Settings*** ➜ ***DDI Settings*** menu

Codecs In the ***VoIP*** ➜ ***Accounts*** ➜ ***ADD/EDIT*** ➜ ***Advanced Settings*** ➜ ***Codec Settings*** menu, you can define which codecs may be used for the selected account:

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP][PROV][ADD][SIP][CODEC]: Profile Settings              MyGateway

           Sorting order           default
           Packet Size in ms       30
           Echo cancellation       on
           Comfort noise           on

 Available Codecs:

<x> G.711 ulaw <x> G.711 alaw  <x> G.729     < > G.726-24
< > G.726-16   < > G.726-32    < > G.726-40  <x> DTMF OoB



            SAVE                           CANCEL

```

Note

The codecs actually used are the intersect of the codecs defined here and those signalled by the provider. For outgoing calls, any remaining codecs are dropped from the list that would require more than the available bandwidth.

The menu contains the following fields:

Parameter	Value
Sorting order	Determines the order in which the codecs are offered for use by the media gateway. If the first codec cannot be used, the second is tried and so on. Possible values: ■ *default* (default value): the codec in the first position in the menu shown above will be used if possible. ■ *quality*: the codecs are sorted by quality. If possible, the codec with the best quality is used. ■ *lowest bandwidth*: the codecs are sorted by required bandwidth. If possible, the codec with the lowest bandwidth requirement is used. ■ *highest bandwidth*: the codecs are sorted by required bandwidth. If possible, the codec with the highest bandwidth requirement is used.
Packet Size in ms	Indicates how much speach data in milliseconds the RTP data packet contains. Possible values: *10* .. *60*. Default value: *30*.

Parameter	Value
Echo Cancelation	Echo cancellation Technique to suppress echo feedback in voice communication on full duplex lines. Possible values: ■ *on* (default value): echo feedback is suppressed. ■ *off*: echo feedback is not suppressed.
Comfort noise	Comfort Noise Generation (CNG) For digital voice transmission, ***COMFORT NOISE GENERATION*** introduces a low level of background noise to avoid the impression that, during pauses at the other end, the connection is lost. Possible values: ■ *on* (default value): CNG is used. ■ *off*: CNG is not used.

Parameter	Value
Available Codecs	Voice coding standard. Possible *values*: ■ *G.711 ulaw*: ISDN codec with US law ■ *G.711 alaw*: ISDN codec with EU law ■ *G0.729*: compressed from 31 to 8 kbit/s; good voice quality ■ *G.726-16*: compressed from 39 to 16 kbit/s ■ *G.726-24*: compressed from 47 to 24 kbit/s ■ *G.726-32*: compressed from 55 to 32 kbit/s ■ *G.726-40*: compressed from 63 to 40 kbit/s ■ *DTMF OoB*: DTMF outband. At first, RFC 2833 will be tried. If the remote device cannot apply this standard, SIP Info will be used. You can enable or disable the codecs shown individually.

Table 2-6: Fields in the ***VoIP → Accounts → ADD/EDIT → Advanced Settings → Codec Settings*** menu

Extensions

In the ***VoIP → Extensions → ADD/EDIT*** menu, you can configure the subscriber numbers of the terminal devices connected to the media gateway, i.e. the subscriber numbers of the SIP terminals and the subscriber numbers of the ISDN or analogue terminals, depending on the available PSTN interfaces.

The menu changes according to the ***Extension Type*** used and the available PSTN interfaces.

For ***EXTENSION TYPE*** = *SIP*.

```
R1200w Setup Tool               Funkwerk Enterprise Communications GmbH
[VOIP][EXT][ADD]: Extension Configuration                     MyGateway


      Number

      Extension Type            SIP

      Advanced Settings >




            SAVE                             CANCEL

```

For ***EXTENSION TYPE*** = *Digital*.

```
R1200w Setup Tool               Funkwerk Enterprise Communications GmbH
[VOIP][EXT][ADD]: Extension Configuration                     MyGateway


      Number

      Extension Type            Digital




            SAVE                             CANCEL

```

The menu contains the following fields:

Parameter	Value
Number	For *Analog* and *Digital*: Subscriber (extension) number. Für *SIP*: User ID. Maximum number of characters: 40.
Extension Type	Terminal device type. Possible values: ■ *SIP*: a SIP terminal device is used for the call. ■ *Digital*: can only be selected if ***ISDN SWITCH TYPE*** = *Euro ISDN point to multipoint (NT Mode)* is set to make ISDN interfaces available. An ISDN terminal device is used for the call.

Table 2-7: Fields in the ***VOIP*** ➜ ***EXTENSIONS*** ➜ ***ADD/EDIT*** menu

In the ***VoIP* ➜ *Extensions* ➜ *ADD/EDIT* ➜ *Advanced Settings*** menu, you make the SIP protocol settings.

```
R1200w Setup Tool                Funkwerk Enterprise Communications GmbH
[VOIP][PROV][ADD][SIP][CODEC]:  Profile Settings              MyGateway



         Auth ID
         Password

         Transport                UDP
         Port                     5060

         Registration Mode        on
         Expire Time              60

         Codec Settings >


             SAVE                                  CANCEL

```

The submenu contains the following fields:

Parameter	Value
Auth ID	Authentication Identifier Here you can enter a name that is to be used for authentication. The name given here must also be entered on the SIP telephone. Maximum number of characters: *20*. If you do not enter a name, the ***NUMBER*** is used.
Password	Enter a password here. The password given here must also be entered on the SIP telephone. Maximum number of characters: *20*.
Transport	Connection protocol to be used Possible values: ■ *UDP*: Data are transferred with UDP (default value). ■ *TCP*: Data are transferred with TCP (default value).
Port	TCP or UDP port number to be used for the connection. Default value: 5060

Parameter	Value
Registration Mode	Enables or disables the SIP REGISTER registration mechanism. Normally, every SIP client (user) sends its current position to a REGISTRAR server by means of a REGISTER message. This information about the user and his current address is held by the REGISTRAR server and queried by other proxies to find the user. Possible values: ■ *on* (default value): the registration mechanism is enabled. ■ *off*: the registration mechanism is disabled.
IP Address	Only for ***Registration Mode*** = *off* For configurations without registration (e.g. connection to a Microsoft Exchange Communication Server) the connection can be setup as static host. Enter the static IP address of the terminal here.
Expire Time	Only for ***Registration Mode*** = *on* Time in seconds after which the current registration becomes invalid and a new registration request is therefore sent. Possible values: *0 .. 38400*. Default value: *60*.

Table 2-8: Fields in the ***VoIP*** ➜ ***Extensions*** ➜ ***ADD/EDIT*** ➜ ***Advanced Settings*** submenu

Codec Settings In the ***VoIP*** ➜ ***Extensions*** ➜ ***ADD/EDIT*** ➜ ***Advanced Settings*** ➜ ***Codec Settings*** menu, you can select the possible codecs for the account. (You will find the possible codecs under "Codecs" on page 29).

PBX In the ***VoIP*** ➜ ***PBX Configuration*** menu, you can configure the exchange functions of the media gateway:

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP][PBX]: Mediagateway Configuration                      MyGateway


    PBX Settings:

         Session Border Controller   OFF
         Media stream termination    on
         Dial latency (sec.)         5
         Default Extension


         Line setup (ISDN) >
         Call Routing >
         CLID Setup >
         Address Translation >
         Speed Dialing >



                   SAVE                           CANCEL

```

The menu contains the following fields:

Parameter	Value
Session Border Controller	Determines the behaviour of the media gateway in combination with a session border controller. Possible values: ■ *AUTO*: for all extensions that exactly agree with an existing account, the call routing is handled by the session border controller, i.e. all SIP messages configured for the corresponding account are forwarded to the session border controller. For all other extensions, the call routing is handled by the media gateway in accordance with the configured call routing entries. Note that the call routing is handled by the media gateway if the provider is not available (backup). ■ *SIP trunk account*: a SIP trunk account is configured and selected for the session border controller under ***VoIP*** ➜ ***Accounts***. In this case, the call routing for all extensions is handled by the session border controller, all SIP messages are forwarded to the session border controller. Note that the call routing is handled by the media gateway if the provider is not available (backup).

Parameter	Value
Session Border Controller (continued)	■ *off*: call routing is handles exclusively by the media gateway in accordance with the configured call routing and the local extensions. For calls that are to be routed via a particular provider (account), you must configure a corresponding call routing entry. Internal calls (from internal extension to internal extension) that are only to be routed internally do not require an additional call routing entry. Note: Call Routing Entries have higher priority than the Session Border Controller configuration!

Parameter	Value
Media stream termination	Determines how RTP sessions are controlled by the system. Possible values: ■ *on*: RTP sessions are terminated on the media gateway, i.e. all RTP streams are controlled by the media gateway and routed via the media gateway. The participating terminal devices (e.g. SIP telephones) are not connected directly with one another. Note that, for VoIP to VoIP connections, there is no code translation for different VoIP terminal codecs. The codecs of media gateway and VoIP terminals must therefore agree. ■ *off* (default value): RTP sessions are not terminated on the media gateway, i.e. all RTP streams are routed by the media gateway without termination. The RTP data packets can be routed in complex networks and thus also via other gateways.
Dial latency (sec.)	Maximum delay time before the system assumes the telephone number entered is complete and starts the SIP dialling process (sends the SIP INVITE message). This timeout is reset each time that a button is pressed. Default value: *5*. If you terminate the number entered with #, dialling is immediate.
Default Extension	Here you can nominate an extension to receive incoming calls that cannot be routed to any extension nor a connected PBX.

Table 2-9: Fields in the ***VoIP*** ➜ ***PBX Configuration*** menu

The ***VoIP* ➜ *PBX Configuration* ➜ *Line setup (ISDN)* ➜ *ADD/EDIT*** submenu appears only if your device has at least two ISDN point to point connections configured in TE (party line) or NT mode.

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP][SIP][LINE][EDIT]: Line                                MyGateway


              Description

              Mode                     trunk

         Available Interfaces
         <x> bri2-1      <x> pri2-4



                        SAVE                         CANCEL

```

Note

Note that, for BRI connections, the connection mode (NT mode or TE mode) must be set by jumper in the device.

The menu contains the following fields:

Parameter	Value
Description	Here you give the party line a name. Maximum number of characters: *40*.
Mode	Mode in which the party line is to be operated. Possible values: ■ *trunk*: PP NT connection (for connection to a PBX). ■ *external*: Telekom party line.
Available Interfaces	For the value in the ***MODE*** field, shows the available ISDN connections. You can enable the ISDN connections to be used for the party line.

Table 2-10: Fields in the ***VOIP*** ➜ ***PBX CONFIGURATION*** ➜ ***LINE SETUP (ISDN)*** ➜ ***ADD/EDIT*** menu

In the ***VoIP → PBX Configuration → Call Routing → ADD/EDIT*** submenu, you can define the conditions for the routing of calls:

```
R1200w Setup Tool               Funkwerk Enterprise Communications GmbH
[VOIP][PBX][CALL][ADD]: Routing                               MyGateway



         Description
         Admin Status            enable
         Type                    external

         Calling Line (opt)      any
         Calling Address (opt)
         Called Address

     Order Line             Rule            Status
   ------------------------------------------------------



        SAVE           ADD          DELETE         CANCEL

```

The menu contains the following fields:

Parameter	Value
Description	Here you give the call routing entry a name.
Admin Status	Determines whether the entry is enabled. Possible values: ■ *enable* (default value): the entry is used. ■ *disable*: the entry is not used.
Type	Determines how calls are to be routed. Possible values: ■ *external* (default value): For calls that are to be routed as outgoing, external calls. This can be done using standard SIP accounts or SIP trunking accounts in DDI client mode. ■ *trunk*: for calls to be routed to a PBX or an ISDN TE connection or a SIP DDI client behind the Media Gateway that is connected by a PSTN line. For this, the following can be used: – PRI interfaces in NT mode, – BRI interfaces in NT mode, – SIP trunking accounts in DDI server mode. ■ *deny*: for calls that are not to be routed (to be blocked).

Parameter	Value
Calling Line (opt)	Here you can restrict the routing entry to the line on which the call comes in. Possible values: ■ *PRI*: restricts the routing entry to the selected PRI interface. ■ *BRI*: restricts the routing entry to the selected BRI interface. ■ *FXO*: restricts the routing entry to the selected FXO interface. ■ <*SIP Account*>: restricts the routing entry to the selected SIP account. ■ *any*: no restriction of the routing entry.
Calling Address (opt)	Here you can restrict the routing entry to a particular caller. To do this, you must specify the subscriber number exactly (no wildcards).
Called Address	Here you can enter an address numerically (e.g. a subscriber number) or alphanumerically (e.g. for a trunk) that is to be compared with a dialled address. The following wildcards can be used: * means that at the end of a character string an arbitrary number of any characters can follow, ? is a placeholder for an arbitrary character. If the configured address agrees with the signalled address, the routing entry is used.

Parameter	Value
Rule	Only for ***Type*** = *trunk* The rule determines how the subscriber number is manipulated before it is used for dialling. Notation: <a:b>, i.e. a is replaced by b. A number of rules can be chained together using semicolons as separators, e.g. <a:b>;<c:d>;<e:f>. After confirmation of entry, the rule chain is automatically sorted by the "best match" method. Numerical and alphanumerical values are permissible. ? is a placeholder for an arbitrary character. Example: rule: <:+49911> number dialled: 96731234 manipulated number: +4991196731234
Line	Only for ***Type*** = *trunk* Defines the line used for an outgoing call.

Table 2-11: Fields in the ***VoIP*** ➜ ***PBX Configuration*** ➜ ***Call Routing***➜ ***ADD/EDIT*** menu

In the ***VoIP*** ➜ ***PBX Configuration*** ➜ ***Call Routing*** ➜ ***ADD/EDIT*** ➜ ***ADD/EDIT*** submenu, you can create a list of rules that are assigned to the currently selected Call Routing Entry and are used to manipulate the signalled destination subscriber number:

```
R1200w Setup Tool              unkwerk Enterprise Communications GmbH
[VOIP][PBX][CALL][ADD][ADD]: Backup Route                    MyGateway


         Admin Status            enable

         Order                   1

         Line                    Not assigned
         Rule



                SAVE                         CANCEL

```

The menu contains the following fields:

Parameter	Value
Admin Status	Defines if the filter rule is to be applied. Possible values: ■ *enable* (default value): the entry is used. ■ *disable*: the entry exists but is not used.
Order	Determines the order of the filter rules, starting with *1* in increasing numerical order. The rules are worked through in the order given in the list. If a line or SIP account is not available, the next rule is automatically used. The corresponding integer for the next rule is shown automatically when it is created. The following criteria disable a line or an account: ■ ISDN – ISDN LAYER 2/1 fault, caused, for example, by a cable fault – ISDN error codes 0x26: network error 0x22: channel not available 0x03: no route to transit network 0x29: temporary error 0x2A: network busy (overload) 0x3A: bearer service not available

Parameter	Value
Order (continued)	■ SIP – SIP registration error caused by a faulty configuration of the SIP account. – SIP registration failed because the provider could not be reached, e.g. because of a DSL problem. – the call monitoring found that the bandwidth of the uplink interface was insufficient for this connection. – SIP error codes: 415: media type not supported 409: code conflict 502: defective gateway 503: service not available
Line	Defines the PSTN line (PRI, BRI, FXO) or the SIP account used for an outgoing call.
Rule	The rule determines how the subscriber number is manipulated before it is used for dialling. Notation: <a:b>, i.e. a is replaced by b. A number of rules can be chained together using semicolons as separators, e.g. <a:b>;<c:d>;<e:f>. After confirmation of entry, the rule chain is automatically sorted by the "best match" method. Numerical and alphanumerical values are permissible. ? is a placeholder for an arbitrary character. Example: rule: <:+49911> number dialled: 96731234 manipulated number: +4991196731234

Table 2-12: Fields in the ***VoIP* ➜ *PBX Configuration* ➜ *Call Routing* ➜ *Add/Edit* ➜ *Add/Edit*** menu

In the ***VoIP*** ➜ ***PBX Configuration*** ➜ ***CLID Setup*** ➜ ***ADD/EDIT*** submenu, you define the processing of the calling party number for incoming calls. You can, for example, add a prefix to a received telephone number in order to route corresponding outgoing calls via a particular account.

```
R1200w Setup Tool                Funkwerk Enterprise Communications GmbH
[VOIP][PBX][CLID][ADD]: Assignment                             MyGateway

         Description

         Calling Line           bri2-2

         Called Line (opt)      any
         Called Address (opt)

         Rule                   <:99>

            SAVE                             CANCEL
```

The menu contains the following fields:

Parameter	Value
Description	Here you give the entry a name.
Calling Line	Here you select the PSTN line (PRI, BRI, FXO) or SIP account from which the call comes.
Called Line (opt)	Here you have the option of entering the destination line of the call (PRI, BRI, FXO or SIP account). Enter either ***CALLED LINE (OPT)*** or ***CALLED ADDRESS (OPT)***. If a value is selected that is not *any*, ***CALLED ADDRESS (OPT)*** should not be used. If you set ***CALLED LINE (OPT)*** = *any* and ***CALLED ADDRESS (OPT)*** is not used, all calls on the ***CALLING LINE*** are processed.
Called Address (opt)	Here you have the option of entering the destination address of the call. Enter either ***CALLED LINE (OPT)*** or ***CALLED ADDRESS (OPT)***. If ***CALLED ADDRESS (OPT)*** is used, you should set ***CALLED LINE (OPT)*** = *any*.

Parameter	Value
Rule	Transformation rule to be used on the subscriber number Notation: <a:b>, i.e. a is replaced by b. A number of rules can be chained together using semicolons as separators, e.g. <a:b>;<c:d>;<e:f>. After confirmation of entry, the rule chain is automatically sorted by the "best match" method. ? is a placeholder for an arbitrary digit. Example: rule: <:+49911> Subscriber number. 96731234 manipulated number: +4991196731234

Table 2-13: Fields in the ***VoIP*** ➜ ***PBX Configuration*** ➜ ***CLID Setup*** ➜ ***ADD/EDIT*** menu

In the ***VoIP*** ➜ ***PBX Configuration*** ➜ ***ADDRESS Translation*** ➜ ***ADD/EDIT*** submenu, you can create a list for the translation of subscriber numbers, i.e. this list associates internal and external numbers.

Note

Which number (called party number or calling party number) is translated depends on the direction (incoming or outgoing) of the call in question. For incoming calls it is the called party number, for outgoing calls the calling party number that is translated.

For example, the internal number 340 can be shown externally as 09119673900 or a call from outside for the number 09119673200 can be routed internally to the number 340.

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP][PBX][ADDR][ADD]: Translation                         MyGateway

         Description

          Direction              both

         Local Number

         Assigned to
         Remote Number


            SAVE                             CANCEL

```

The menu contains the following fields:

Parameter	Value
Description	Here you give the number translation a name.
Direction	Here you enter the direction to which the entry is to apply. Possible values: ■ *both* (default value): for incoming and outgoing calls (bidirectional) ■ *incoming*: for incoming calls ■ *outgoing*: for outgoing calls
Local Number	Here you enter the internal number (e.g. extension or PBX number). For incoming calls, the signalled Called Party Number (corresponds in the menu to the ***REMOTE NUMBER*** field) is translated to the ***LOCAL NUMBER***. For outgoing calls, the signalled Calling Party Number (corresponds in the menu to the ***LOCAL NUMBER*** field) is translated to the ***REMOTE NUMBER***. Numerical and alphanumerical characters are permissible. ? is a placeholder for an arbitrary digit. Note ***LOCAL NUMBER*** and ***REMOTE NUMBER*** must contain the same number of wildcards.
Assigned to	Determines the line (PRI, BRI, FXO, FXS) or SIP account via which the calls are to be routed.

Parameter	Value
Remote Number	Here you enter the external (e.g. ISDN MSN or SIP account subscriber number). For incoming calls, the signalled Called Party Number (corresponds in the menu to the ***REMOTE NUMBER*** field) is translated to the ***LOCAL NUMBER***. For outgoing calls, the signalled Calling Party Number (corresponds in the menu to the ***LOCAL NUMBER*** field) is translated to the ***REMOTE NUMBER***. The ***REMOTE NUMBER*** is not shown if the field setting is ***ASSIGNED TO*** = <*SIP Account*>. In this case, the ***USER ID*** of the selected SIP provider account is used as the ***REMOTE NUMBER***.

Table 2-14: Fields in the ***VOIP*** ➜ ***PBX CONFIGURATION*** ➜ ***ADDRESS TRANSLATION*** ➜ ***ADD/EDIT*** menu

In the ***VOIP*** ➜ ***PBX CONFIGURATION*** ➜ ***SPEED DIALLING*** ➜ ***ADD/EDIT*** submenu, you can define speed dial numbers for frequently used subscriber numbers.

```
R1200w Setup Tool                Funkwerk Enterprise Communications GmbH
[VOIP][PBX][CALL][ADD]: Speed Dialing                          MyGateway


         Description

         Shortcut number
         Replace to



            SAVE                               CANCEL

```

The menu contains the following fields:

Parameter	Value
Description	Here you enter a description for the user.
Shortcut Number	Here you enter the desired speed dial number for the user, e.g. *123*.
Replace to	Here you enter the subscriber number to be dialled in place of the speed dial number, e.g. *09119673*. In the example above, if a user types in **123*, the device dials *09119673*. If the user wishes to call extension 111, he types in **123111*. The device dials *09119673111*. A period at the end of the number indicates a complete number. This is dialled immediately when the period is recognised.

Table 2-15: Fields in the ***VoIP*** ➜ ***PBX Configuration*** ➜ ***Speed Dialing*** ➜ ***ADD/EDIT*** menu

If you wish to use a speed dial number from the list under ***VoIP*** ➜ ***PBX Configuration*** ➜ ***Speed Dialing***, you must press * and then the speed dial number.

Dynamic Bandwidth Control

For telephone calls over the Internet, VoIP packets normally have the highest priority. Nevertheless, if the upstream bandwidth is low, noticeable delays in voice transmission can occur when other packets are routed at the same time. The Dynamic Bandwidth Control function in the VoIP implementation solves this problem. So as not to block the "line" for VoIP packets for too long, the size of other data packets is reduced if need be during a telephone call.

Note

If you use an external modem and wish to use dynamic bandwidth control, you must state the bandwidth of the upload connection.

Note

If you wish to use an external modem, then you must set the fields ***QUEUEING AND SCHEDULING ALGORITHM*** = *priority queueing (PQ)* and ***SPECIFY TRAFFIC SHAPING*** = *yes* in the ***QOS*** ➜ ***INTERFACES AND POLICIES*** ➜ ***EDIT*** ➜ ***QOS SCHEDULING AND SHAPING*** menu.

The configuration is set in the ***VOIP*** ➜ ***DYNAMIC BANDWIDTH CONTROL*** ➜ ***ADD/EDIT*** menu.

```
R1200w Setup Tool              Funkwerk Enterprise Communications GmbH
[VOIP][LFI][ADD]: Configure Jitter Reduction                 MyGateway

  Interface         1000    en1-0


  Mode               enabled for all RTP data

  Maximum Link Speed in
  Upload Direction (bit/s)   10000000


  Please specify really available Upload Speed      0



            SAVE                           CANCEL
```

The menu contains the following fields:

Parameter	Value
Interface	Here you select the upload connection on which the voice transmission is to be optimised.
Mode	Determines the optimisation mode. Possible values: ■ *enabled for all RTP data* (default value): By means of RTP data, the system recognises VoIP data traffic and optimises the voice transmission. This setting should be used only if the media gateway is not enabled (DSP module not installed) or not used. ■ *enabled for controlled RTP data only*: By means of the data routed through the media gateway, the system recognises VoIP data traffic and optimises the voice transmission. This setting should always be used together with the media gateway. ■ *always*: Voice data transmission is always optimised. ■ *disabled*: Voice data transmission is not optimised.
Maximum Link Speed in Upload Direction (bit/s)	Shows the maximum bandwidth of the upload connection.
Please specify really available Upload Speed	If you use an external DSL modem, you must enter the bandwidth.

Table 2-16: Fields in the ***VOIP*** ➜ ***DYNAMIC BANDWIDTH CONTROL*** ➜ ***ADD/EDIT*** menu

3 Problems Solved

The following problem has been solved in System Software 7.6.6:

3.1 Setup Tool - Irrelevant menus shown

(ID 10077)

The Setup Tool displayed the menus for Pre and Post IPSec Rules even in the case of purely interface based configurations. A configuration in these menus could then lead to unexpected results.

Attention!

Generally, it is not recommended to configure any Pre or Post IPSec Rules in a completely interface based IPSec configuration.

In case you have Pre- or Post IPSec Rules defined, these will remain effective after the update. In order to delete them (if necessary) you can create a temporary peer on the basis of traffic lists. The respective menus will the be accessible again, and you can delete any undesired rules. Afterwards you should also delete the temporary peer again.