Documentation

on

SIP PBX trunking

with SIP-DDI 1.0

and the related

QSC product

IPfonie®extended

and the QSC solution module

IPfonie®extended VPN

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List of references

Author	Document
Roland Hänel	"Technical Specification of SIP-DDI 1.0"
Andreas Steinkopf	“Specification of IPfonie®extended VPN” and “Specification of IPfonie®extended VPN”

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1 Introduction

The replacement of the POTS ("plain old telephone system") by Internet telephony has made significant progress in Germany and other European countries. QSC AG (in the following "**QSC**") started operations in this area using "SIP PBX trunking" or "SIP-DDI" on the 16th August 2006.

"SIP PBX trunking" means the connection of a telephone system and unified messaging and media servers (in the following referred to as a private branch exchange [**PBX**]) to the public telephone network via VoIP. SIP-DDI 1.0 is the SIP variant, which QSC defined with other major German IP-PBX manufactures in their official journal. It is based on the ITU Q.1912.5. SIP trunking means that from an external point of view, the telephone service behaves like a conventional telephone line where connection possibilities and service features are concerned, in spite of the use of VoIP technology. The advantage is that companies can continue to use their existing PBX systems and gradually shift towards Internet telephony.

Using "SIP PBX trunking" or SIP-DDI 1.0, Internet telephony receives all the relevant service features of ISDN. In particular this includes Direct Dial In (**DDI**), which allows individual telephone numbers (DDI numbers) to be assigned to extension lines or departments, so that the caller can call these extensions directly without having to connect to the switchboard first. The fact that DDI was not previously available with Internet telephony was considered one of the main obstacles hindering the use of SIP throughout entire telephone systems, from the individual extension through to the public switching exchange.

The familiar service features of ISDN primary rate interfaces are maintained, from dynamic CLIP evaluation (forwarding of the calling number by the terminal device) through to fax functionality. Unlike a primary rate interface, SIP PBX trunking allows the system capacity (number of possible simultaneous uplink voice channels) to be selected not only in steps of 30 voice channels, but also in steps of 10.

Physically, QSC has supplied this new type of connection in the form of the standard product "IPfonie®extended" – for use in the public Internet – and the QSC solution module "IPfonie®extended VPN" – for use in the virtual private networks (**VPN**) supplied by QSC – since 16 August 2006.

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2 Network Components

QSC has installed a complete VoIP platform based on an NGN, which covers all the possibilities for using VoIP.
This network includes the following components:

- Soft switches
- Media gateways
- Session Border Controller (configured back-to-back user agent, **B2BUA**)
- SIP servers
- VoIP management system
- If operated in a QSC-IP-VPN, a NAT breakout to the NGN

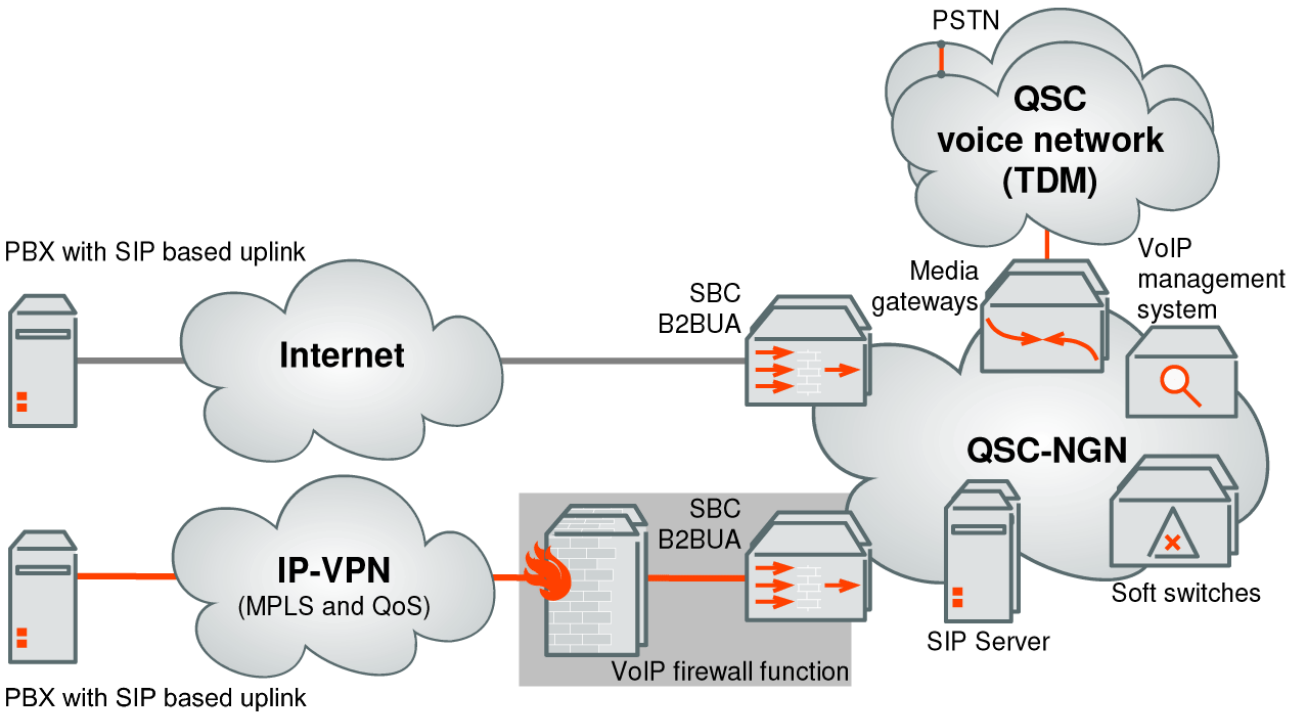

Figure 1: Simplified schematic diagram of the QSC VoIP platform

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3 SIP PBX trunking

3.1 Reasons for SIP PBX trunking

With SIP PBX trunking (QSC product name: “IPfonie®extended”), it is possible to model all the essential service features of ISDN on a PBX basis. In particular this includes so-called Direct Dial In (DDI), which allows individual telephone numbers (DDI numbers) to be assigned to extension lines or departments, so that the caller can call these extensions directly without having to connect to the switchboard first. The fact that DDI was not previously available with Internet telephony was considered one of the main obstacles hindering the use of SIP throughout entire telephone systems, from the individual extension through to the public switching exchange.

Direct Dial In (DDI) Support:
DDI means the direct calling of a specific person’s line from a telephone network. This feature allows defined individual extensions to be dialled directly. In order to support DDI, the respective SIP provider must be able to assign multiple telephone numbers to the same SIP account or SIP login (with login name and login password).

3.2 The General Principle behind SIP PBX trunking

Normally, SIP implies a fixed relation between the SIP connection and a corresponding public telephone number. SIP PBX trunking dissolves this relationship and allows additional dialling characters to be appended to the root number. This is referred to as “prefix binding” in the following.

Example:
A SIP account has been registered with the root number “02216698”. Now if, for example, a connection is to be established to the number “02116698-123”, the INVITE message is forwarded via the SIP connection “02216698” (“prefix binding”).

3.3 Call Setup – Preparation

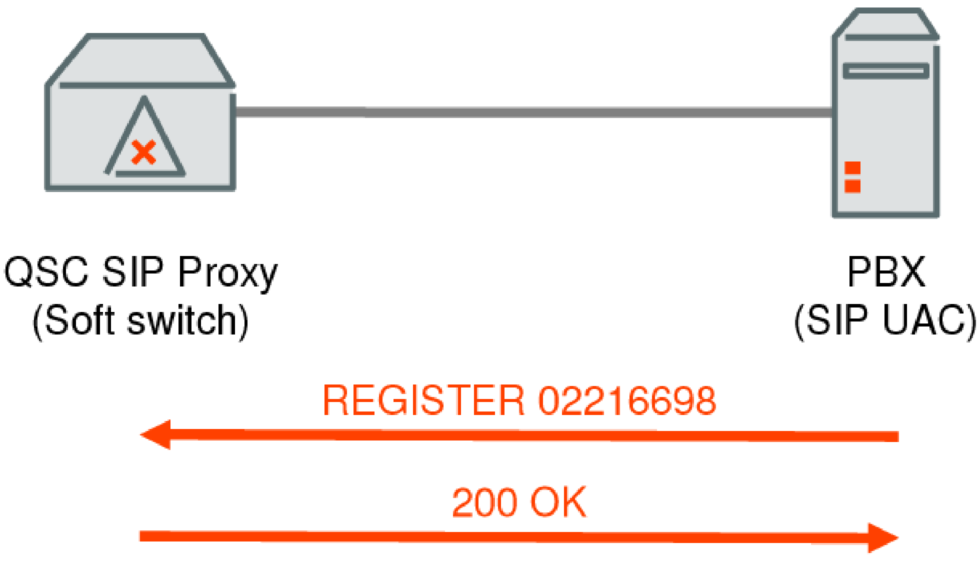

Figure 2: Call Setup – Preparation

The PBX (e.g. the IP PBX, soft PBX or the media or UMS server), which assumes the role of the SIP UAC (User Agent Client), registers the prefix “02216698” via the QSC SIP proxy, thus creating a dynamic assignment for each PSTN number that begins with the prefix “02216698” and that might have additional dialling characters assigned via the SIP proxy (“prefix binding”).

The REGISTER request for the case of SIP PBX trunking (SIP message) is thus in no way different from a normal REGISTER request as used for other SIP connections. Here, the functionality of “prefix binding” is solely an internal function of the SIP proxy and has no other effect on the SIP protocol. This means that neither a new structure nor changes to the SIP header are necessary.

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3.4 Call Setup – Incoming Call

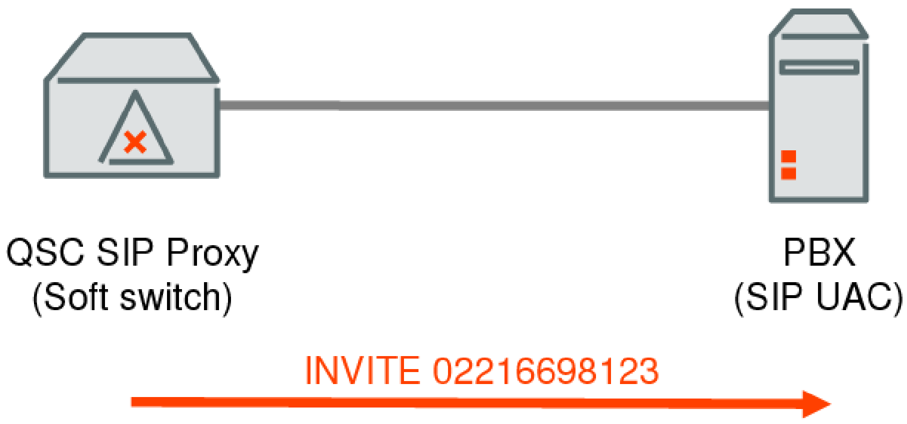

Figure 3: Call Setup – Incoming Call

Here, each INVITE message (e.g. from the PSTN) that can be attributed to the “prefix binding” (root number) is forwarded according to its assignment.

In this respect, it is important to note that the additional dialling characters (extension numbers) of the IP PBX are only communicated via the “To:" header” and the "X-ORIGINAL-DDI-URI:- header" of the incoming INVITE, because the Request URI is assigned to the “prefix binding”.

In the case of a call deflection to the SIP DDI account, the redirect number can be entered in the "To:" header. Therefore, for new implementations, it is recommended that only the "X-ORIGINAL-DDI-URI:- header" is used for new implementations.

In addition to the "X-ORIGINAL-DDI-URI", the "P-Called-Party Header" was implemented. This adopts the same function as the previously used X-ORIGINAL-DDI-URI:- header. Both headers are used in parallel to achieve backwards compatibility. The "P-Called-Party" header is described in RFC 3455 chapter 4.2.

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Example: normal call flow

Root number
Root number + extension
Root number + extension

```
INVITE     sip:02216698@2.2.2.2;user=phone SIP/2.0
FROM:      sip:030123456@sip.qsc.de;user=phone>;tag=908e7475
TO:   <sip:02216698123@2.2.2.2;user=phone>

X-ORIGINAL-DDI-URI:sip:02216698123@sip.qsc.de;user=phone
P-Called-Party-ID: <sip:02216698123@sip.qsc.de:5060>;user=phone
[…]
```

Example: Redirect from 0160 12345678

Root number
Redirect
Root number + extension

```
INVITE sip:02216698@2.2.2.2:10897 SIP/2.0
From: <sip: 0301234561@sip.qsc.de;user=phone>;tag=17e95717
To: <sip:16012345678@2.2.2.2;user=phone>
[…]
X-ORIGINAL-DDI-URI:sip:02216698123@sip.qsc.de;user=phone
P-Called-Party-ID: <sip:02216698123@sip.qsc.de:5060>;user=phone
```

3.5 Call Setup – Outgoing Call

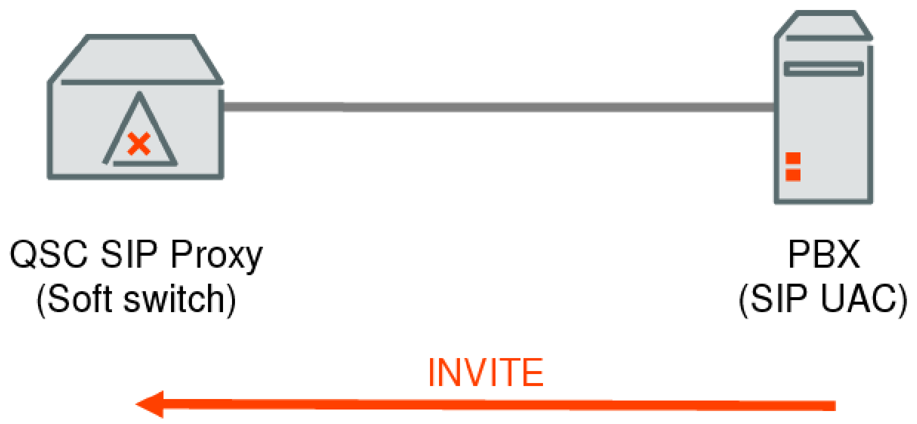

Figure 4: Call Setup – Outgoing Call

Here, each INVITE message (e.g. from the IP PBX) that can be attributed to the “prefix binding” is forwarded according to its assignment. It is important to consider that the additional dialling characters of the IP PBX are only communicated via the “From:” header of the outgoing INVITE, because the SIP URI is assigned to the “prefix binding”.

Example:

```
From: <sip:02216698123@213.148.135.2;user=phone>;tag=c9f84ba5
To: <sip:0301234561@213.148.130.46;user=phone>
CSeq: 1 INVITE
[…]
```

3.6 ClipNoScreening

In order to use the *ClipNoScreening* function with an outgoing call, the P-Asserted-Identity is inserted in the INVITE message. The calling number belonging to the account must be transferred in the P-Asserter field. If this calling number matches the account, the call is forwarded, otherwise the INVITE is declined with the message "`403 Only valid users are allowed in INVITE PAI`".

If a valid P-Asserted-Identity exists, a user provided A-number can be transferred in the FROM header.

The mapping in ISUP messages appears as follows:

P-Asserted-Identity ⇨ Network provided Number
FROM Header ⇨ User Provided (Generic Number)

It should be noted here that different carriers may have different displays for the user provided number.

If no P-Asserted header is available, the call number in the FROM header is checked for valid account data and forwarded accordingly.
Thus the function ClipNoScreening is backwards compatible, as SIP INVITE messages without the P-Asserted Header are handled as previously.

Example:

```
INVITE sip:02212922999@1.2.3.4 SIP/2.0
Via: SIP/2.0/UDP 9.8.7.4:5061;branch=z9hG4bK2227675c
From: "Testcall" <sip:0800999999@9.8.7.4>;tag=as419dfad3
To: <sip:02212922625@1.2.3.4>
Contact: <sip:02212925719@9.8.7.4>
Call-ID: 123456789@9.8.7.4
CSeq: 102 INVITE
User-Agent: Test
Max-Forwards: 70
Date: Tue, 15 Jul 2008 10:51:52 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
P-Asserted-Identity: <sip:022129257190@sip.qsc.de:5060;user=phone>
Content-Type: application/sdp
Content-Length:   263
```


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3.7 Additional Notes on IPfonie®extended

The QSC VoIP platform currently supports the voice codecs G.711 and G.729a. The limitation of the number of simultaneous VoIP calls (concurrent calls) depends on the voice codec used as well as the available bandwidth of the transmission path. Therefore, a limitation of the "concurrent calls" must be configured on the connected IP PBX.

Please note that QSC uses "CLI screening" for outgoing calls (INVITE from the IP PBX to the SIP proxy).

- "CLI screening" allows extensions for the PBX users. For outgoing calls, it is possible to append additional characters. For example, the message "From: <sip:02216698123@..>" will be accepted, as it is successfully authenticated via the SIP account "02216698".
- The authentication – via MD5 digest – always refers to the corresponding SIP account. In respect of the above example, this means that "02216698" is always used as the "user name", in the WWW authenticate, even if the IP PBX uses "02216698123" as the telephone number.

SIP PBX trunking – and thus IPfonie®extended – does not implement additional PBX functions that could be provided via the QSC voice network.

- For SIP (Session Initiation Protocol), QSC supports the RFC3261 standard. Extended SIP methods such as SUBSCRIBE, OPTIONS, NOTIFY and REFER are not supported by QSC. If required, these must be implemented directly on the IP PBX, for example a second outgoing call in the case of call deflection (CD).
- QSC supports the use of DTMF tones according to the RFC2833 standard (RTP payload for DTMF digits).
- QSC supports fax transfer via the T.38 protocol. For this protocol, which currently is not yet identically implemented in all cases, testing and approving of the function with the respective IP PBX is highly recommended as is implementation using IPfonie®extended.
- The calling number format used is E.164. It is represented as "user info" in the SIP URI.
- QSC supports CLIP / CLIR (transmission / suppression of the calling number). The calling party number is provided in the "From" field in compliance with RFC3261, chapter 8.1.1.3. The number specified in this field is transmitted (CLIP). To suppress calling number transmission (CLIR), the "From" field is filled with "Anonymous".

In case partial calling number blocks are to be routed to the IP PBX, it is necessary to use several "prefix bindings".

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Example:
The number range from 02216698-100 to 02216698-399 is to be assigned to the IP PBX.

- The first “prefix binding” is then 022166981.
- The second “prefix binding” is 022166982.
- The third “prefix binding” is 022166983.

IMPORTANT: In this case, the IP PBX must be capable of handling these three SIP bindings, since three separate REGISTER requests from three independent SIP accounts must be supported. Each of these “prefix-bindings” must be treated as explained above in this document.

4 Objectives for T.38 equipment

4.1 Transfer of CNG and CED tones

As both fax machines and other terminal units (e.g. fax switches in answering machines, etc.) exist on the market which require flawless CNG- (calling tone) and CED- (called terminal identification) tones, an, as far as possible, interference-free transmission of these tones is necessary. A basic principle is that the tonal signalling during phase A of the T.30 transmission must remain as unchanged as possible.

One approach would be to identify the fax connections as quickly as possible using the CNG tone, to switch over to T.38 and to transfer the CNG and CED tones via T.38 using T.30 indications. This approach has numerous disadvantages:

- The sending of CNG and CED T.30 indications is optional in T.38. I.e. this approach would not work with many ATAs.
- Reliable and robust fax identification is only possible using V.21 flags. A switchover which is as fast as possible due to the presence of CNG or CED tones runs the risk of incorrect switchovers.
- Tone identification with subsequent switchover leads, at least in all the implementations tested up until now, to at least a strongly garbled tone, the identification of which on the partner side is uncertain.

A second approach would be to transfer the CNG and CED tones as RTP events in the audio channel. This approach has the following disadvantages:

- The method is unusual and is rarely supported by gateways or ATAs.
- The alternating transmission of RTP packets and RTP events leads to garbled tones at the partner with severe amplitude oscillations because the quick and clear identification of CNG and CED tones can only be achieved with great difficulty.

In practice, a third approach has proven successful:
The CNG and CED tones are transmitted as RTP audio data and only when the V.21 flag is recognized does transmission switchover to T.38. This makes a clean interruption-free tone transmission possible. If speech compression is being used, there is indeed a slight change in tone. However up until now, this has not had a negative effect in practice as the recognition tolerances of both tones are specified with relatively large tolerances (CNG +-38 Hz, CED +-15 Hz). During switchover to T.38 after identification of the V.21 flag and a relatively short preamble to the V.21 datagram, garbling of a V.21 datagram can occur. This is not a problem as the T.30 standard clearly specifies that only datagrams with a correct

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CRC can be evaluated and in practice all fax machines conform to this specification. If, when using this approach, the partner switches over to T.38 earlier, then the CNG and CED tone are naturally to be transferred as T.30 indications.

ATAs and gateways should therefore be correspondingly configured and tested so that they

- transmit CNG and CED in the RTP audio data up to the fax machine (listening test necessary!)
- switchover to T.38 upon detection of a V.21 flag on the TDM side
- transmit CNG and CED T.30 indications in the event of early switchover
- to not transmit CNG and CED as RTP events (Fax CNG, Fax ANS)

4.2 T.4 ECM (Error Correction Mode)

As the T.38 standard does not specify ECM as either optional or mandatory, there are some T.38 implementations which do not support ECM and which, by means of a manipulation of the T.30-DIS (digital identification signal) message prevent the fax machines from using ECM.

Without T.4 ECM, fax machines are generally not capable of correcting errors in analogue transmission using page transmission via V.17, V.29 or V.27. This means that in any event, occasionally incorrect page transmissions should be allowed for, dependent on line quality and the quality of the modem algorithm (in the fax machines, ATAs and gateways).

The support of T.4 ECM by all components (fax machines, ATAs, gateways) is an absolute necessity in a professional environment. As from experience there are some ATAs which implement ECM incorrectly, it could nevertheless, dependent on the client, make sense to disable ECM via the gateway.

4.3 Modulation for page transmission

The support of V.17 (with 14400 and 12000 bit/s) enables accelerated page transmission in comparison with the transmission via V.29 (9600 Bit/s). Admittedly both for gateways as well as for ATAs testing should at least be carried out with a few common commercial fax machines in order to determine which transmission rates can actually be achieved.

If the quality of the modem algorithm of the T.38 machines is so poor that the fax machines must process the data at slower data rates, then the transmission duration increases markedly. An ATA that reliably supports V.29 or even only V.27 and signals this accordingly, is nevertheless considerably better than an ATA which does indeed signal V.17 but whose training then fails 8 times.

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4.4 Redundancy

If packet loss is expected for a particular IP-path (including routers and LANs which are the customer's responsibility), then the redundancy mechanism of the T.38 standard should be configurable for both V.21 signalling and also page transmission. A triple redundancy for V.21 messages and a quadruple redundancy for the T.38 page transmission, as is currently configured on the QSC VoIP platform on pages of the Huawei gateway, appears reasonable and should also be configurable on the ATAs used.

In so doing, broadband limitations should be borne in mind. The maximum useful data transfer rate for V.21 messages is equal to 300 bit/s x redundancy factor. The maximum useful data transfer rate for page transmission is equal to the maximum modulation rate (e.g. 14000 bit/s) x redundancy factor. The overhead due to the header is dependent on the packet size. Typical packet sizes for page transmission are between 20 and 40 ms. T.38 packet sizes are not explicitly configurable with any known existing fax machines. In special cases with limited bandwidth, configurable packet lengths could make sense.

If packet loss is to be minimised by prioritisation, it must be checked whether the measure is also effective in conjunction with the T.38 protocol.

4.5 Packet Delay Variation resp. Jitter

If considerable Packet Delay Variation (PDV) resp. jitter is expected for a particular IP-path (including routers and LANs which are the customer's responsibility), then the gateway-ATA combination must be tested under appropriate conditions. PDV of 150 ms leads to considerable problems in the tests carried out. The PDV settings of the ATAs and gateways do not, from experience, relate to the T.38 transmission, so that if PDV problems exist, it appears likely that the manufacturer should be contacted.

If PDV is to be minimised by prioritisation, it must be checked whether the measure is also effective in conjunction with the T.38 protocol.

4.6 re-INVITE collisions

If both sides of a T.38 transmission implement the same type of fax identification, there is, dependent on the type of fax identification, a realistic risk that due to line echoes, both machines will identify a T.30 fax simultaneously and switch the connection over to T.38.

The collision case of two re-INVITEs is described in RFC3261. Both sides must answer with a status 491 (request pending) and after a random time between 10 ms and 2 s or 4 s repeat the re-INVITE, if the cause of the re-INVITE still exists. This method is suitable for switching over to T.38, as the called fax only repeats its DIS datagram after 3 s.

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However, the method is not correctly implemented by all gateways and ATAs. As the collision is caused by line echoes, the probability of an error is dependent on the actual individual construction. This makes field errors extremely problematic.

Therefore it is an absolutely necessity to check for the correct implementation of the re-INVITE collisions in the ATAs and gateways and provoke such a collision in the test. One possibility for increasing the possibility of a collision, is to select a connected fax machine from an Asterisk solution using ISDN via the gateway of the ATA under test and in so doing to start the Asterisk echo application. Now the ATA and gateway on the TDM-side should see the same signal, irrespective of the unavoidable delay, so that the probability of a collision is optimised.

4.7 Port numbers

Both in respect of NAT as well as in respect of "independent-minded" T.38 variants (Cisco), it is appropriate that the port numbers of the T.38 connection are chosen to be identical to the preceding RTP connection. The selection of the T.38 port should therefore be checked and, as necessary, the manufacturer of the T.38 product, overruled accordingly.

4.8 Parallel signalling of T.38 and "clear channel" / "fax pass through"

The parallel transmission of T.38 and "clear channel" creates room for interpretation and possible error sources. A sequential signalling of both options is preferable. Should parallel signalling be encountered on T.38 devices, the manufacturer should be pressured into providing a sequential variant.

4.9 T.30 No signal indications

So that NAT sessions can be opened in the event of new port numbers and so that incorrect T.38 stacks can be persuaded to cooperate together, T.30 no signal indications make sense, particularly at the start of the T.38 session. If a gateway or ATA does not send T.30 no signal indications at the start of the session, even though no signal exists, then this behaviour should be recommended to the manufacturer.

4.10 DTMF

Neither ATA nor gateway should send DTMF/telephone events as RTP events within a T.38 session. It must also not be announced within the SDP parameter. ATAs and gateways should ignore such events and their announcement in the SDP and not break off the connection.

Within the framework of voice communication, DTMF/telephone events should be transmitted in both directions (i.e. not only from the calling party to the called party, but also

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in the opposite direction) and signalled in the SDP, so that services based on callback are also enabled as in "normal" telephone networks.

4.11 RTCP

Incoming RTCP packets should, within the framework of the RTP session, be correctly terminated and generate their own reports as debug aids. After completion of the RTP session, no RTCP packages should be generated for this session.

4.12 Special software

In general, ATAs are relatively easy to update with new software, even at the customer's premises. Therefore, there is a realistic possibility of the manufacturer being able to rectify the detected problems by the provision of special "QSC software" and then use of this software at customers in a targeted manner in order to achieve a quality lead relative to other providers. Inherent in this variant is of course an increase in support overheads, even if this only involves the importing of new software at the end customer.

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5 Remarks about other PBX functions

5.1 Limiting of the number of voice channels at distributed locations

In many cases, the PBX (e.g. an IP-PBX) is installed in a central customer location or at the carrier ("hosted PBX"), the VoIP terminals (VoIP telephones and softphones), which are connected to it, however, are located in other locations (distributed branches, subsidiaries, home working locations).

These distributed locations are connected with the PBX via IP over Internet or IP-VPN links. As the bandwidth available for voice data is not unlimited but rather has a specified maximum value according to assiduous planning, only a limited number of VoIP voice connections can be made simultaneously via Internet and IP-VPN links.

Example: the customer location is linked with a 2 Mbit/s IP-VPN link. It has been agreed with the customer that 40 % of this bandwidth should ensure that 8 G.711 voice channels are available. By use of DiffServ and Class of Service (for example with QSC using the option "QSC®Class of Service") the minimum bandwidth of 800 kbit/s is reserved for the voice data.

If during operation, the planned or set bandwidth for voice connections is exceeded in that more than the planned number of voice connections are made simultaneously over the IP-VPN link, then packet loss occurs and consequently a dramatic worsening of voice quality.

This means that the PBX (or the VoIP application layer) must be configured for the number of simultaneous voice connections which can be made to or from a location (site) and that the number of voice channels is robustly monitored and limited in accordance with this configuration for each location. To clarify this, this only refers to voice channels which leave the location and not voice channels which are made between extensions within the same location.

5.2 Limiting of the number of voice channels at the SIP central office

That the above described voice channel number must be configured and limited also applies to the PBX SIP central office, as even for the PBX IP connection only certain, plannable bandwidths are available for voice data.

This can for example be implemented in that over a specifiable "SIP trunk" instance which may be made up of a multiplicity of SIP accounts, this SIP trunk can be allocated a maximum number of voice channels.

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6 FAQs

6.1 Does SIP trunking using IPfonie®extended also work without registration?

Answer:
Essentially no.

The reasons are mainly:
without registration would mean that the carrier/provider would have to configure the receivers. I.e. every PBX at the end customer would have to be rigidly configured with its own IP address (or with that of its NAT gateway!) by the carrier/provider. This would make all redundancy scenarios difficult and create a high operating overhead.

Apart from that, in general an end customer SBC (at QSC: Huawei manufacturer or Acme packet) does not support any "unregistered accounts". For SIP trunks without registration there are other SBCs (or firmware for this), which however are aimed at the VoIP wholesale or carrier market.

Therefore, operation without registration does not make any sense for the end customer product IPfonie®extended. In would then be an actual SIP trunk, no longer an account. In principle an SIP trunk would work excellently, however it is a high overhead for voice routing (comparable with the setting up of #7-E1 couplings). Likewise considerable license costs would arise from it within the NGN. Therefore, such an SIP trunk is rather a carrier/provider solution, which does not however make sense for the connection of a certain number of end customer PBXs.

IPfonie®extended is exactly the correct approach for modelling certain features, which with normal SIP only exist on a trunk, in an SIP account.

Conclusion: the IP-PBX manufacturers will open up a new market for themselves once they can implement an economic end customer central office SIP-DDI. Many well known manufacturers have clearly identified this and are currently in the process of implementing SIP-DDI.

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6.2 Further typical questions from manufacturers

Q1: What Layer 4 protocols are currently supported and are they mandatory or optional? (UDP, TCP, TLS, SCTP)
A1: Only UDP is supported using SIP.

Q2: What Layer 4 protocols are planned for future releases and when are they likely to be rolled out?
A2: Currently, no further L4 protocols are planed

Q3: What Uri's types are supported? (SIP Uri, Tel Uri without phone context, Tel Uri with phone context)
A3: QSC supports only SIP Uri with phone context.

Q4: Does the service require the support of P-RACK
A4: PRACK (Provisional Response ACKnowledgement) is supported, but is not required for connecting SIP devices.

Q5: What voice codec's are supported?
A5: G.711 aLaw, G.729 annex B.

Q6: Does network provide ring back or does it require this to be locally provided. Does QSC require early media capability (180/183 with SDP) in the CPE SIP endpoints?
A6: QSC does not provide ring back, it should be provided locally. We did not require early media in the SIP endpoints.

Q7: What type of authentication used i.e. digest, with or without "qop=auth-int", etc?
A7: QSC only supports the digest authentication scheme (MD5) using "qop=auth".

Q8: What services/features are provided by the network, that requires feature support on the CPE in order to interoperate e. g. Message waiting indication, hold, transfer…
A8: The following network features are supported for SIP: CLIP, CLIR, COLP, call waiting and call hold. Any further features must be provide by the CPE

Q9: Can your registrar handle multiple contacts with different q values? Can a particular contact be removed through unregistration leaving all other registrations intact?

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A9: The QSC registrar is able to handle multiple contacts with different q-values. It is also possible to remove a particular contact and to leave all other registrations intact

Q10: Do you envisage operation behind a NAT? If so, do you provide SBC/ALG functionality for endpoints to traverse through NAT?
A10: QSC is using a SBC which supports network based NAT traversal.

Q11: How many numbers can you allocate per trunks?
A11: No technical restrictions are known for allocating the quantity of numbers per trunk.

Q12: Do you use refresh on an existing call (RFC4028, support for session timers)? Do they support UPDATE (RFC3311)?
A12: RFC3311 is supported. RFC4028 is currently not supported.

Q13: Do you support Fax? If so, can Fax be offered at the beginning of the session? Or does fax transmission occur only in the middle of a call?
A13: Currently QSC supports T.38 only with the Re-Invite method. If the media gateway detects a fax tone, it sends again an Invite with 'UDP transport layer t38'.

Q14: Do you support Invite with no SDP?
A14: QSC supports Invite messages without a SDP part.

Q15: If you support G.729 do you support Annex B or Annex A or both? Do you send and accept attribute for Annex B or Annex A in SDP?
A15: QSC supports only G.729 annex B.

Q16: Do you support RFC 2833 for DTMF tones?
A16: RFC 2833 is supported for the transport of DTMF tones.

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