Artist - SIP Configuration Application Guide V1.1 (AP-S0100)

Thank you for choosing a RIEDEL product.

This Application Note helps you to configure the setup of the SIP environment for the Artist system.

For further information please read the Director User Guide.

The Artist VoIP-108 firmware can act as SIP Phone by registering as an extension to a SIP conform Private Branch Exchange (PBX) such as Avaya Session Manager, Cisco Call Manager, 3CX, Freeswitch and Asterisk. The Artist system provides control panel key commands to place or answer a telephone call to other PBX extensions. There is no need for assigning the remote PBX extensions only to SIP phones; depending on the capabilities of your PBX, the call can be routed to a non SIP extension e.g. a FXO extension or a PSTN subscriber.
Every single port is able to register at a SIP registrar. Hence, using a VoIP-108-G2 card, eight simultaneous calls are feasible.

Generally the VoIP-108 G2 performs the role of a SIP User Agent Client (UAC), which sends SIP requests, and performs also the role of the User Agent Server (UAS), which receives the requests and returns a SIP response. These roles of UAC or UAS only last for the duration of a SIP transaction, so during a single telephone call.

SIP also defines server network elements. Although two SIP endpoints can communicate without any intervening SIP infrastructure, which is why the protocol is described as peer-to-peer, this approach is often impractical for a public service or a large private environment. RFC 3261 defines those server elements like Proxy and Registrar Server. Typically today's conventional private branch exchange system (PBX) are offering additional to the automatic switching for the subscriber also the SIP Register and or the Proxy server.

Proxy server
An intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients. A proxy server primarily plays the role of routing, which means its job is to ensure that a request is sent to another entity "closer" to the targeted user. Proxies are also useful for enforcing policy (for example, making sure a user is allowed to make a call). A proxy interprets and, if necessary, rewrites specific parts of a request message before forwarding it.

Registrar
A server that accepts REGISTER requests and places the information it receives in those requests into the location service for the domain it handles which registers one or more IP addresses to a certain SIP URI, indicated by the SIP: scheme, although other protocol schemes are possible (such as tel:). More than one user agent can register at the same URI, with the result that all registered user agents will receive a call to the SIP URI.
SIP registrars are logical elements, and are commonly co-located with SIP proxies. But it is also possible and often good for network scalability to place this location service with a redirect server.

Within the PBX, the administrator must create and configure a “SIP User” or in a general term an “Extension”. Because in most cases the VoIP-PBX has adapted the “old” telephone user interface, an extension is still a short number, e.g. 3 or 4 digits for speed dial. This Extension ID is mapped to the SIP Username within a SIP URI.
Within a SIP Header it is also possible to define an optional display name. The display name can transport further information, e.g. the full name of the user. The SIP User Agent can display this in addition to the Username.
To secure the access of a registration to the SIP server the SIP RFC 3261 defines authentication methods for the communication. The VoIP-108-G2 supports HTTP and Proxy Authentication.

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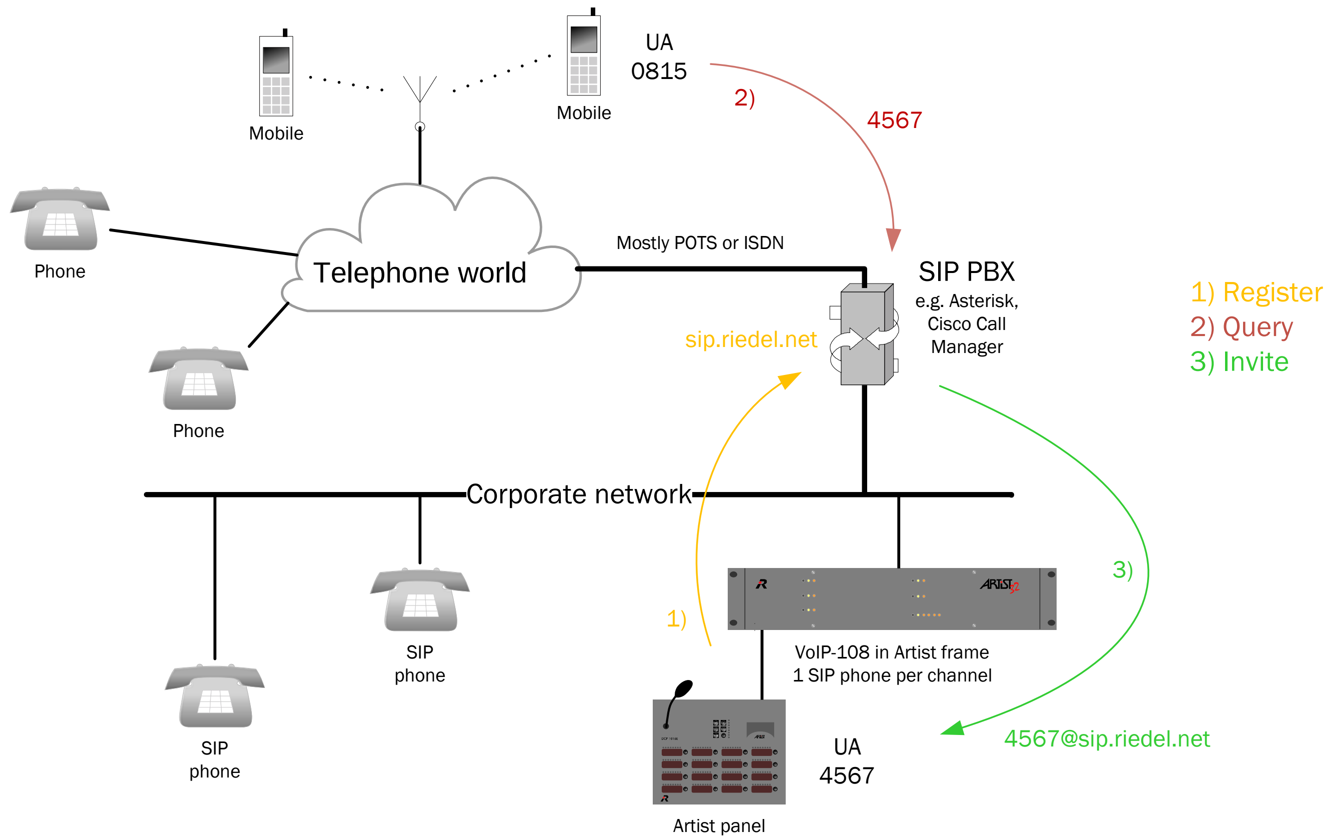
Setup example:



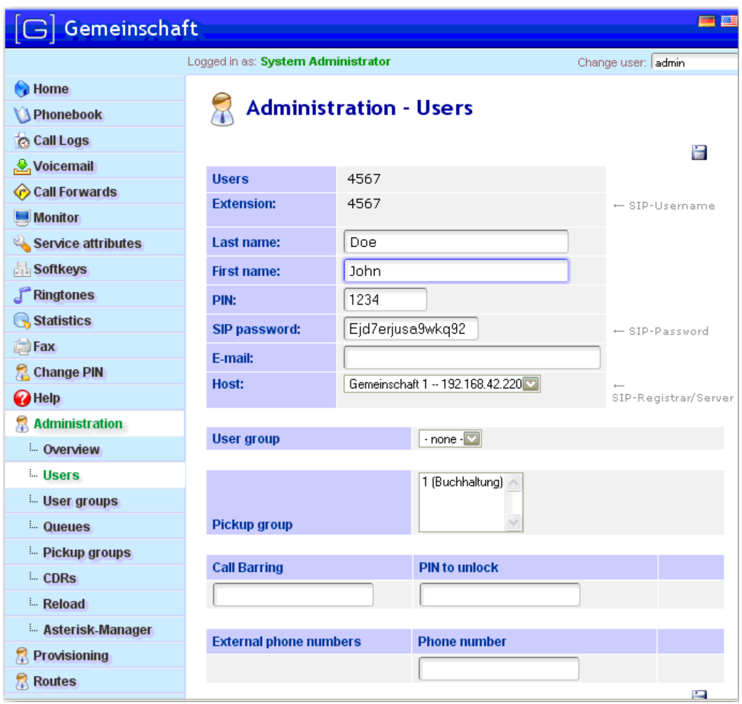
Example for a SIP extension 4567 configuration in the web-interface of an Asterisk based PBX:

At PBX “Gemeinschaft” site:

The extension is 4567, the full user name is “John Doe” and the SIP PBX is hosted on “sip.riedel.net”:

- Username / Extension: 4567
- Display Name: “John Doe”
- Authentication Username: 4567
- Authentication Password. Ejd7erjusa9wkq9233

Resulting SIP Header Field “To”:
“John Doe” <sip:4567@sip.riedel.net>.



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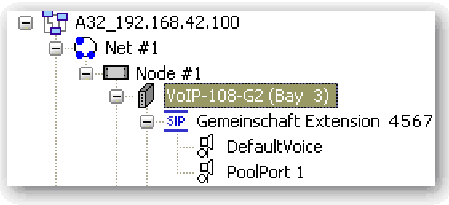
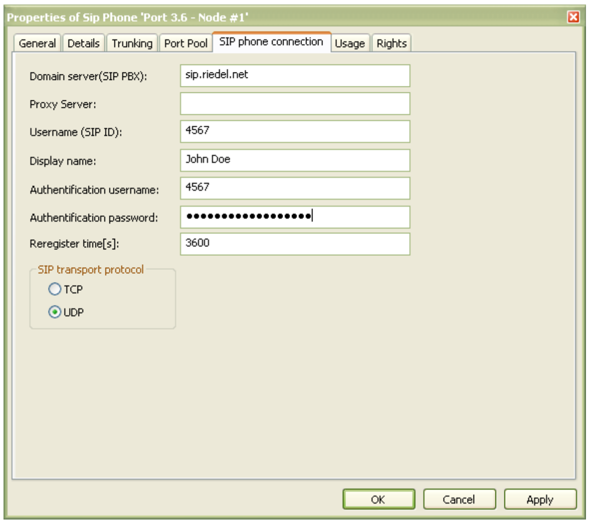
At Artist site:

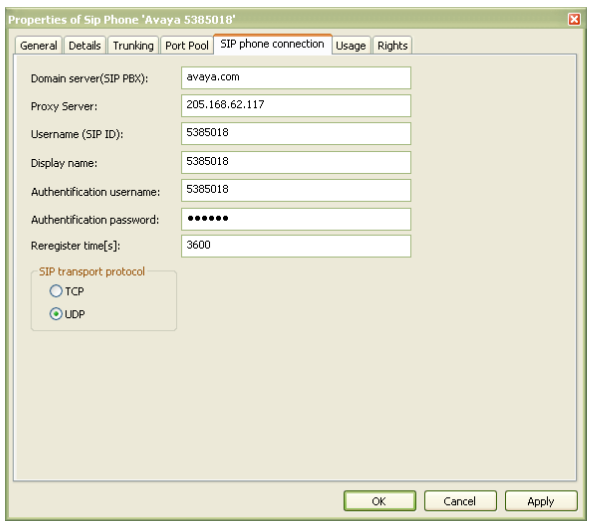
For a SIP REGISTER request to a server these parameters are also needed on the side of the SIP User Agent, so the VoIP-108-G2 card in the Artist frame. The configuration on Artist side is done with the Director Software. SIP Phone Connections can be configured on any port of a VoIP-108-G2 card. For further information on the Riedel VoIP card please refer the Artist Installation manual and the Director user guide.



Parameters for a SIP registration must be configured in the properties page of a SIP Phone Connection. The SIP PBX in our example is “sip.riedel.net” via DNS, otherwise a plain IP address of the PBX is needed.



If the PBX-System is using a Proxy server, the VOIP-108G2 must send all requests first to the proxy to pass all messages to the final Register server.
For this case, the properties of the SIP Phone Connection which can be placed on a VOIP-108G2 offers a further text field for the proxy server address.



VoIP-108 G2 SIP Specification

Riedel's implementation follows the current RFCs for performing SIP and handling RTP, RTSP, SDP and SAP with SIP.

- RFC 3261: The Basic SIP Protocol (http://www.ietf.org/rfc/rfc3261.txt)
- RFC 4566: SDP: Session Description Protocol (http://www.ietf.org/rfc/rfc4566.txt)
- RFC 3264: SDP Offer/Answer Negotiation (http://www.ietf.org/rfc/rfc3264.txt)
- RFC 3551/3550: RTP, Real-Time Transfer Protocol (http://www.ietf.org/rfc/rfc3551.txt)

Supported SIP Methods are:

- REGISTER
- OPTIONS
- INVITE
- ACK
- CANCEL
- BYE

For SIP, TCP and UDP are supported on a selectable port, the standard port is 5060.

For the coded RTP Audio Payload, G.711 and G.722 are provided.

RTP / RTCP are provided via UDP

Mandatory parameters for the registration of the VoIP-108-G2 card on a SIP Server:

- Username (SIP ID): This is the user name inside the PBX domain.
- Display name: This name of the port is visible for others, deviant from the Username.
- Authentication User name: This name is used for login at the SIP server.
- Authentication Password: Password for the login.
- UDP or TCP for establishing the connection to the domain server.

More specific information can be found in the Director manual and in your PBX manual.