RTS

APPLICATION NOTES:
SIP Server

PROBLEM/SOLUTION

Intercom systems often have a need to connect to phone systems. In the past, Plain Old Telephone Service, or POTS, was readily available from the local phone company, but with the migration to IP, old-style POTS lines are hard to get. For a while, ISDN lines seemed to be the future, but in most locations, even ISDN is disappearing. Session Initiation Protocol, or SIP, is the dominant IP-based protocol for telephony today. SIP is the protocol spoken by most Private Branch Exchanges, or PBXs. The topic of this Application Note is connecting SIP phones to an intercom.

SIP-SOLUTIONS FROM RTS

RTS has hardware solutions that provide SIP-connectivity. However, we will look at the possibility of providing SIP through a software solution that requires very little additional hardware, and is highly scalable. Figure 1 shows the basic system diagram.

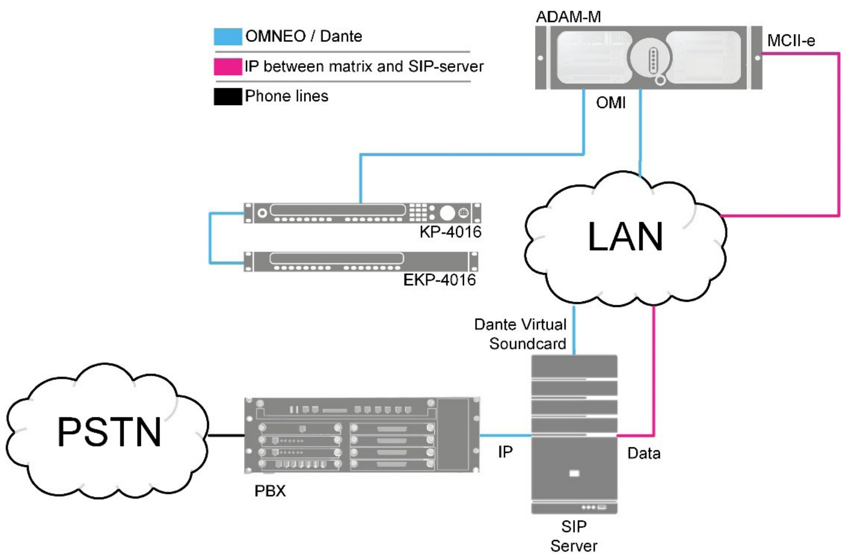

Figure 1. The basic SIP-solution

The matrix, an ADAM-M in this example, is connected to a dedicated PC running special software. The PC acts as a SIP Server. It has two pieces of software installed:

- Dante Virtual Soundcard from Audinate – this is the software device that sends audio to, and receives audio from, the matrix in the form of high-quality audio over IP. The ADAM-M has an OMI-card, which sends and receives OMNEO, the Bosch media networking solution that supports Dante and other protocols.

- VLINK software – this RTS software is responsible for handling the SIP communication itself. SIP is simply a software feature within the VLINK software. SIP can be enabled by simply checking a box in the VLINK setup screen.

Beyond the components of the IT network itself, no other hardware devices are required. The master controller of the ADAM or ADAM-M matrix is able to speak directly to the VLINK software, over IP. The SIP-functionality is supported in the current master controller software (version 3.3.0 and later).

If the matrix does not have an OMI-card, it is possible to use an analog connection. However, since the SIP-Server is expecting digital audio over IP, it must be converted to Dante with an external conversion box as shown in Figure 2. The example uses a Focusrite RedNet 1 box.

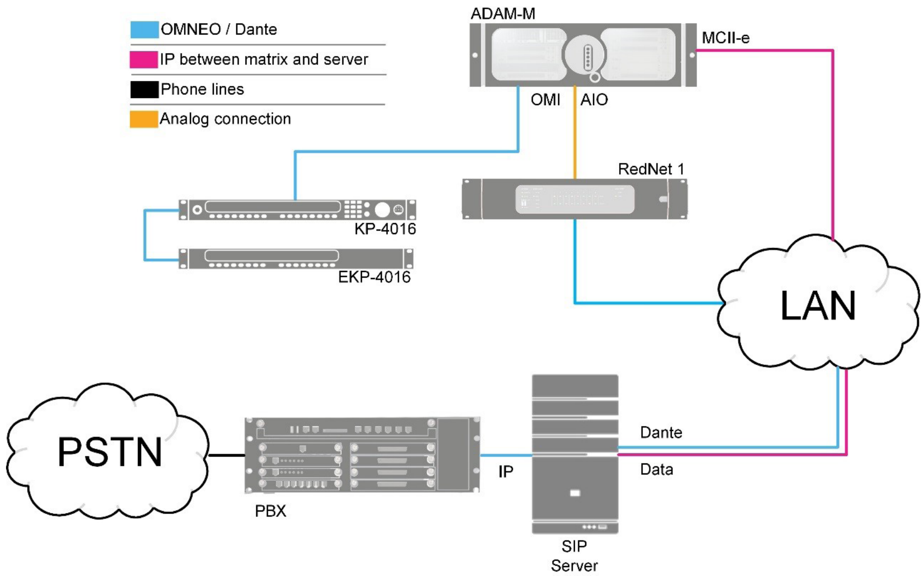

Figure 2. SIP-example using an analog output from the matrix

MIXED USE OF VLINK

As we have seen, VLINK is capable of being an interface to SIP. The traditional use of VLINK for intercom applications, which is still supported, is to run the software on a smartphone and have it work as a keypanel. This requires a separate server, as shown in Figure 3.

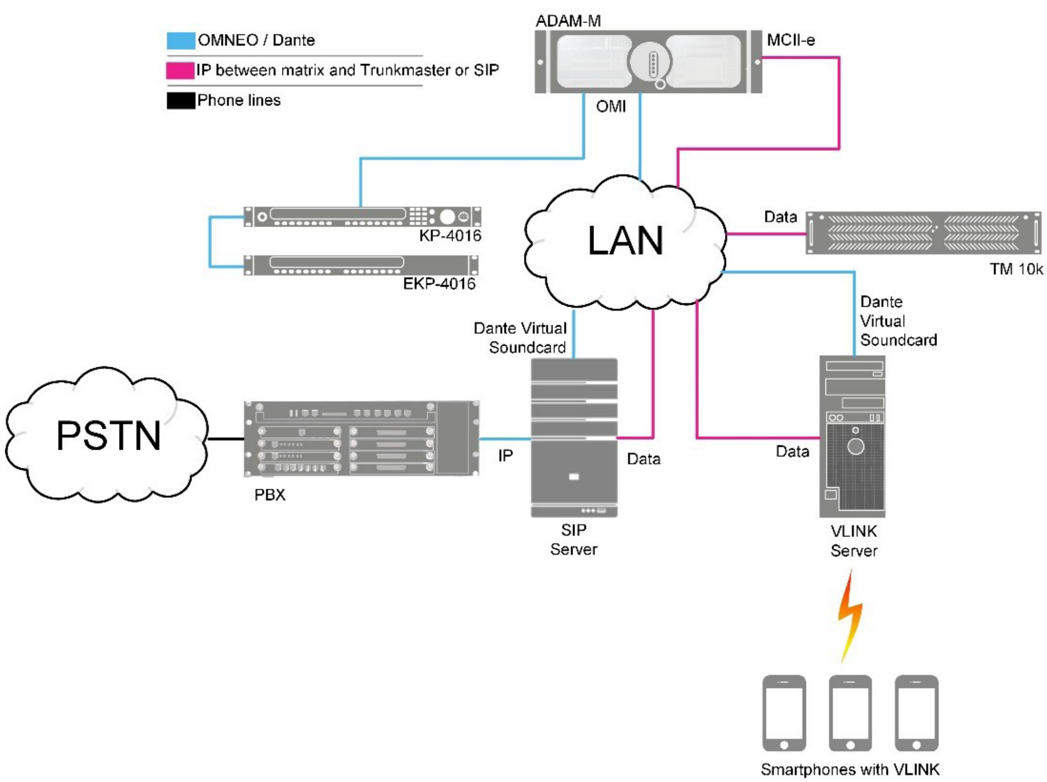

Figure 3. Use of VLINK for SIP and for Smartphones requires two servers

TECHNICAL CONSIDERATIONS

- SIP is available for the ADAM and ADAM-M matrices.
- The SIP-Server requires VLINK licenses, plus a Dante Virtual Soundcard.
- Each SIP line requires two VLINK ports, which must be licensed.
- SIP-Server does not require a Trunkmaster.
- When VLINK is used for SIP and for smartphones, two separate servers are required.
- An OMI-card is recommended in the matrix. If analog audio is used, an external converter must be used.
- A software-based Tally-screen is available separately. It shows the status of all SIP-lines.