Known Issues

LCOS Software Release 7.26

for LANCOM Routers using Voice over IP

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1 Call transfer (connect/transfer)

- **Call transfer fails from/to users on SIP provider lines**

  With various SIP providers, problems can arise when transferring calls (connect/transfer). The reason for this is that REINVITES in the SIP protocol are not fully processed by the SIP providers, so that changes of port, codec or IP address may result in rejection (also see "2 Test of call transfer between various SIP providers")

  Solution: As this behavior can be caused by changes of voice codec, it may help to restrict the available codecs in the settings for the SIP line to the provider. A universal solution would require the LANCOM VoIP Call Manager to be extended with a media proxy. This will be implemented in a future version of LCOS. Use analog or ISDN lines to transfer calls to external subscribers.

- **Two calls between different SIP lines or PBX lines cannot be connected (call transfer**

  Solution: A universal solution would require the LANCOM VoIP Call Manager to be extended with a media proxy. This will be implemented in a future version of LCOS. Use analog or ISDN lines to transfer calls to external subscribers.

- **Two calls with external subscribers over the same SIP-provider line often cannot be connected (call transfer)**

  Solution: The provider has to support call transfer (connect and/or transfer) by means of the REFER method in the SIP protocol. Various providers will be supporting this in future (e.g. QSC). A universal solution would require the LANCOM VoIP Call Manager to be extended with a media proxy. This will be implemented in a future version of LCOS. Use analog or ISDN lines to transfer calls to external subscribers.

- **Call transfer (connect/transfer) fails because of different voice codecs**

  If telephones are operated with differing voice-codec settings, situations can arise where call transfer (connect/transfer) to a local subscriber may not work. This is even more probable when a call is to be transferred from/to a SIP provider or SIP PBX line.

  Solution: Ensure that you are not using different voice codecs in the various telephones or operating differing codec restrictions. For example, set up all devices with the first choice of codec as G.711 µ-law and the second choice as G.711 A-law.

- **Many SIP clients do not return to the original call after the failure of a call transfer**

  After a call transfer has been initiated at a SIP phone, SIP signaling may result in errors or the rejection of the connection request (e.g. as the result of incompatible codec settings). Many SIP phone or clients are then unable to return to the call which is on hold. The call on hold can only be accepted again after hanging up first. This is currently the case for the LANCOM VP-100 and the LANCOM Advanced VoIP Client. Analog and ISDN telephones are generally not subject to this problem because the VoIP Call Manager handles call holding and signaling.

Solution: The SIP devices require improvements that control the SIP signaling and hold the two connections.

- **Generally, unknown problems can occur with SIP phones from other manufacturers**

  Solution: Despite the existing standards, the SIP protocol still has to mature with regard to the support of various services. Many manufacturers continue to rely on proprietary approaches. In combination with the LANCOM VP-100 and LANCOM Advanced VoIP Client, tests have been conducted for the following devices, among others; snom 190, 320, 360, Thomson Speedtouch 2030, Siemens optiPoint 410S, Cisco Unified IP Phone 7912G, Linksys SPA941.

- **Limited support of unattended call transfer**

  With the LANCOM VP-100, a call on hold can be transferred unattended by entering the target telephone number and immediately pressing the "Transfer" key. Analog or ISDN telephones generally do not have a special key or menu option for direct unattended call transfer.

  Solution: A full implementation requires a control mechanism, such as hanging-up while dialing or ringing from analog or ISDN telephones. This will be implemented in a future version of LCOS.

- **Call forwarding (i.e. automatic call transfer) from various SIP devices functions with restrictions only, or not at all.**

  For some SIP phones, pressing the hash key ("#") while dialing with an off-hook handset or with an activated speakerphone function triggers immediate dialing. This means that controlling PBXs with the hash key is either not possible, or it can be used for the first character only.

  The LANCOM Advanced VoIP Client does not signal an asterisk (*) in the SIP-URI when establishing a call if the sequence ends in a hash.

  Solution: Generally speaking, telephone numbers that include control codes (keypad facilities) can be entered into SIP telephones even when the handset is on-hook or when the speakerphone is not active. Picking up the handset or activating the speakerphone then initiates dialing. This method is suitable for controlling PBX functions.

  The VoIP Call Manager now also supports control by this method if the keypad-facility control code does not end with the hash "#" sign. In this way, keypad facilities can be used with the LANCOM Advanced VoIP Client as well.
  Examples

Set up immediate call forwarding to the telephone number 504:

***21*504#** can alternatively be actvated with ***21*504**

Temporarily disable immediate call forwarding:

#*22*# can alternatively be set with **#*22***

- **ISDN facilities for automatic call forwarding not supported**

  Automatic call forwarding with ISDN telephones cannot be operated via the ISDN facilities (functional protocol) but by entering keypad facility code sequences instead, e.g. "#21#".

  Solution: An upgrade to include the ISDN functional protocol is planned for a future version of LCOS. Please use the code sequences (keypad facilities) that are described in the documentation.

2 Test of call transfer between various SIP providers

The following table lists the results of thorough tests of the current version of LCOS with various SIP providers. "**OK**" indicates that the test was a success and "**NOK**" that the test was at least partially unsuccessful.

	Hold/swap/**connect external incoming** SIP **call**	Hold/swap/**connect local outgoing** SIP **call**	**Forward** an **external incoming SIP call** to an **ISDN** landline	**Forward** an **external incoming SIP call** to **local** ISDN/analog/SIP **subscribers**
1und1	**OK**	**OK**	**OK**	**OK**
CARPO	**OK**	**OK**	**OK**	**OK**
PBX-Network	**OK**	**OK**	**OK**	**OK**
QSC (Trunk)	**OK**	**OK**	**OK**	**OK**
SIPCALL	**OK**	**OK**	**OK**	**OK**
SIPGATE	**OK**	**OK**	**OK**	**OK**
Strato	**OK**	**OK**	**OK**	**OK**
T-Online*	**NOK**	**NOK**	**NOK**	**OK**
Comment	Call from provider to the LANCOM VP-100 is connected to an ISDN telephone, from there to an analog telephone a/b1, from there to an analog telephone a/b2, from there to an Advanced VoIP Client, from there to a Cisco IP Phone 7912G (SIP)	The LANCOM VP-100 makes an outgoing call via the provider and connects to an ISDN telephone, from there to an analog telephone a/b1, from there to an analog telephone a/b2, from there to an Advanced VoIP Client, from there to a Cisco IP Phone 7912G (SIP)	An incoming SIP call via the provider is automatically forwarded over an external ISDN line to a mobile phone and an ISDN landline number	An incoming SIP call from the provider is automatically forwarded to a local ISDN subscriber, analog subscriber, and SIP subscriber

* T-Online uses different SIP servers that handle signaling in various manners and with software of various versions. A better result may be achieved under different circumstances.