

Dial String Parameter for the  
Session Initiation Protocol Uniform Resource Identifier

Status of This Memo

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Abstract

RFC 3966 explicitly states that 'tel' URIs may not represent a dial string. That leaves no way specify a dial string in a standardized way. Great confusion exists with the SIP URI parameter "user=phone", and specifically, if it can represent a dial string. This memo creates a new value for the user parameter "dialstring", so that one may specify "user=dialstring" to encode a dial string as a 'sip:' or 'sips:' URI.

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## 1. Introduction

A user at a phone often has a limited User Interface, and in some cases, is limited to a 10 key pad (and sometimes a "flash" function with the switchhook). The user enters a series of digits that invoke some kind of function. The entered sequence, called a "dial string", may be translated to a telephone number, or it may invoke a special service. In many newer designs, the mapping between a dial string and a phone number or service URI is contained within the phone (digitmap). However, there are many phones and terminal adapters that do not have internal translation mechanisms. Without a translation mechanism in the phone, the phone must send the dial string in a 'sip:' or 'sips:' URI [RFC3261] to an intermediary that can transform the dial string to a phone number or a service invocation. The intermediary is able to perform this transform provided that it knows the context (i.e., dialing plan) within which the number was dialed.

There is a problem here. The intermediary can apply its transformation only if it recognizes that the user part of the SIP URI is a dial string. However, there is currently no way to distinguish a user part consisting of a dial string from a user part that happens to be composed of characters that would appear in a dial string.

Use of DTMF (dual tone multi-frequency) detectors after the initial number has been dialed is not uncommon. A common function some systems have is to express a string that incorporates fixed time delays, or in some cases, an actual "wait for call completion" after which additional DTMF signals are emitted. For example, many voicemail systems use a common phone number, after which the system expects the desired mailbox number as a series of DTMF digits to deposit a message for. Many gateways have the ability to interpret such strings, but there is no standardized way to express them, leading to interoperability problems between endpoints. This is another case where the ability to indicate that a dial string is being presented would be useful.

## 2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

It is assumed that the reader is familiar with the terminology and acronyms defined in [RFC3261].

### 3. Requirements

A mechanism to express a dial string in a 'sip:' or 'sips:' URI is required. A dial string consists of a sequence of

- \* the digits 0-9
- \* the special characters # and \*
- \* the DTMF digits A-D
- \* characters representing a short pause, and a "Wait for call completion" in a dial string

Note: DTMF = dual tone multi-frequency. Each "tone:" is actually two frequencies superimposed. DTMF is a 4 x 4 matrix with four row frequencies (697, 770, 852, 941 Hz) and four column frequencies (1209, 1336, 1477, 1633). Most telephones only implement 3 of the 4 columns, which are used just as the telephone dial pad implies. Thus, the digit 2 is the first row, second column, and consists of 770Hz and 1209Hz frequencies mixed together. The fourth column is not used in the PSTN (Public Switched Telephone Network). The "digits" for the fourth column are usually expressed using the letters A through D. Thus, "C" is 852/1633Hz. Some systems do use these digits, so we include them in the definition of the dial string.

A dial string always exists within a context. The context MUST be specified when expressing a dial string.

It MUST be possible to distinguish between a dial string and a user part that happens to consist of the same characters.

### 4. Solution

A new alternative value for the "userinfo" parameter of the 'sip:' or 'sips:' URI schemes is defined, "dialstring". This value may be used in a 'sip:' or 'sips:' URI when the user part is a dial string. The dial string is a sequence of the characters 0-9, A-F, P, X, '\*' and '#'. E represents \*, F represents #, P is a pause (short wait, like a comma in a modem string) and X represents "wait for call completion".

When the "user=dialstring" is used, a context parameter, as defined in [RFC3966], MUST be specified. The context parameter would normally be a domain name. The domain name does not have to resolve to any actual host but MUST be under the administrative control of the entity managing the local phone context. The context parameter

value is normally configured in the user agent.

The syntax of the context parameter follows the same conventions as the same parameter in [RFC3966], that is, it appears between the digits and the "@" in the userinfo [RFC3261] of the URI:

```
dialstring = dialstring-digits context; context from RFC 3966
dialstring-digits = *dialstring-element dialstring-digit
                  *dialstring-element
dialstring-digit = HEXDIG / "*" / "#"; HEXDIG from RFC 3966
dialstring-element = dialstring-digit / "P" / "X" /
                  visual-separator; visual-separator from RFC 3966
```

A dial string SHOULD NOT be used for an AoR (Address of Record) in a REGISTER. Parameters are ignored in registration. Thus, two registrations with different phone-contexts would be considered equivalent, which is probably not desirable.

A proxy server or Back to Back User Agent (B2BUA) [RFC3261], which is authoritative for the context, may translate the dial string to a telephone number or service invocation URI. The telephone number MAY be expressed as a global or local tel: URI, or it MAY be left as a sip: or sips: URI with the URI parameter value changed from "user=" to "user=phone".

Examples of dial string use include:

```
;what a SIP Phone might emit when a user dials extension 123
sip:123;phone-context=atlanta.example.com@example.com;user=dialstring
```

```
;existing voicemail systems have a local access extension,
;then expect to see the extension number as DTMF for the mailbox
sip:450X123;phone-context=biloxi.example.com@example.com;user=dialstring
```

## 5. IANA Considerations

[RFC3969] defines a 'sip:' or 'sips:' URI Parameter sub registry. The "user" parameter is specified as having predefined values.

This RFC defines a new value for the "user" parameter, "dialstring". This RFC has been added to the references listed for the "user" parameter.

## 6. Security Considerations

Dial strings exposed to the Internet may reveal information about internal network details or service invocations that could allow attackers to use the PSTN or the Internet to attack such internal

systems. Dial strings normally SHOULD NOT be sent beyond the domain of the UAC (User Agent Client). If they are sent across the Internet, they SHOULD be protected against eavesdropping with TLS (Transport Layer Security) per the procedures in [RFC3261].

## 7. Normative References

- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.
- [RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, June 2002.
- [RFC3966] Schulzrinne, H., "The tel URI for Telephone Numbers", RFC 3966, December 2004.
- [RFC3969] Camarillo, G., "The Internet Assigned Number Authority (IANA) Uniform Resource Identifier (URI) Parameter Registry for the Session Initiation Protocol (SIP)", BCP 99, RFC 3969, December 2004.

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